TELECOMMUNICATIONS
BEG 435 EC

Teaching Schedule Hours/Week			Examination Scheme				
Theory	Tutorial	Practical	Internal Assessment		Final		Total
3	1	3/2	Theory	Practical*	Theory	Practical**	
			20	25	80	-	125

*** Continuous**
**** Duration**: 3 hours
Course objectives: The course objective is to give fundamental of telecommunication system.

1. Introduction **(3 hrs)**
- 1.1 Evolution of telecommunication
- 1.2 Structure of telecommunication system
- 1.3 Simple telephone communication

2. Transmission media **(10 hrs)**
- 2.1 Transmission media characteristics
- 2.2 Transmission line
- 2.3 Twisted pair, Feeder cable and coaxial cable
- 2.4 Microwave principle components and communication
- 2.5 Optical fibre communication

3. Signal Multiplexing **(4 hrs)**
- 3.1 Space division multiplex
- 3.2 Frequency division multiplex
- 3.3 Time division multiplex

4. Switching system **(8 hrs)**
- 4.1 Switching techniques
- 4.2 Space division switching
- 4.3 Time division switching

5. Subscriber and Signaling in telecommunication **(6 hrs)**
- 5.1 Rotary dial telephone
- 5.2 Touch tone dial telephone
- 5.3 Subscriber loop signaling
- 5.4 Interexchange signaling
- 5.5 Intraexchange signaling

6. Data communication and computer networking **(10 hrs)**
- 6.1 Structure of local area networks
- 6.2 Local area network protocols
- 6.3 Network interfaces
- 6.4 Inter-networking
- 6.5 Routine and flow control

7. Telephone traffic and networks **(5 hrs)**
- 7.1 Fundamentals of telephone traffic
- 7.2 Telephone network
- 7.3 Integrated service digital network (ISDN)

Laboratory:
Six laboratory exercises in FDM, TDM, Switching signal transmission in coaxial cable, optical fibre cable, microwave components.

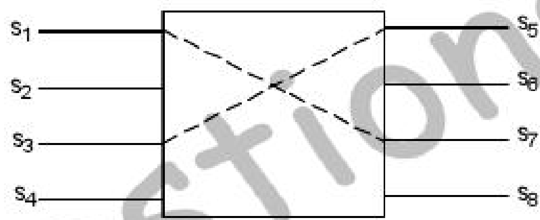
References:

1. M.Schwartz "Telecommunication networks" Addition Wesley.
2. B.E.Briley "An Introduction to telephone switching" Addition Wesley
3. W. Stallings "Local Area Networks" Mc Millan
4. Harold B. Killen "Fibre Optic Communications" Prentice Hall
5. Manuals published by telecom equipment.

Evolution of telecommunication:

Telegraph was introduced in 1837 in Great Britain and in 1845 in France. In March 1876, Elexander Grahm Bell demonstrated his telephone set and the possibility of telephony. His model was based on point to point connection between entities. In general case, there are n(n-1)/2 links with an entities. Network with point to point links among all the entities are known as fully connected network. The disadvantage of such model is that the no of links required becomes very large with moderate values of M, also receives more numbers of cables etc.

To overcome above difficulties, scientists invented concepts of switching system with the introduction of SS the subscriber do not connected to one another but are connected to SS as shown in figure.

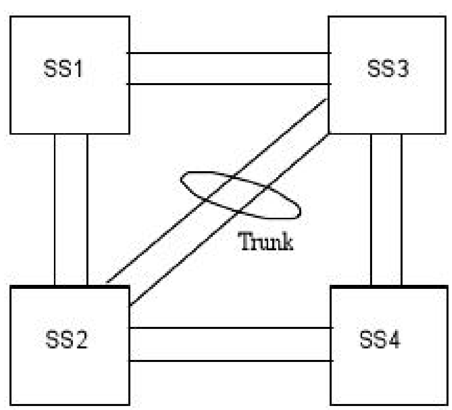


In this figures only one link is required between subscriber hand- switching system and total number of link is equal to the number of subscriber connected to the networks.

Generally, there were number of modification in switching system. Early switching system were manual and operator oriented. Which provided lots of problems in establishing call and was inefficient in time perspective. To overcome above drawbacks manual exchange was replaced by automatic exchange. The automatic exchange have large number of advantages as it require less time to establish and release call; maintain privacy etc.

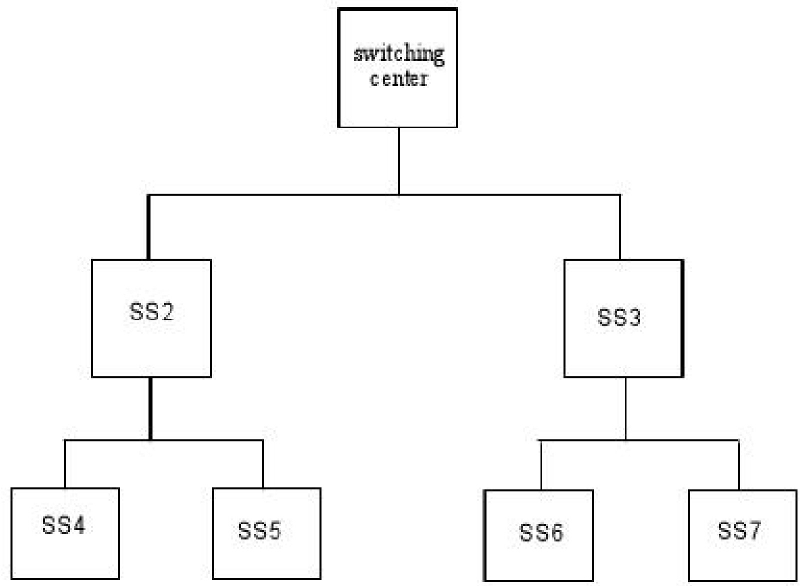
Gradually, the automatic exchange was also modified as electromechanical, electronic, time division switching, space division switching etc.

However, subscriber all over the world cannot be connected to single switching system. To overcome this problem, switching system placed at different geographical location and connected to each other to provide services between different distance subscriber as shown in fig.



The links that run between switching system are called trunk. The no of trunks depend upon the traffic between two switching system as the number of exchange are increased, the number of links and

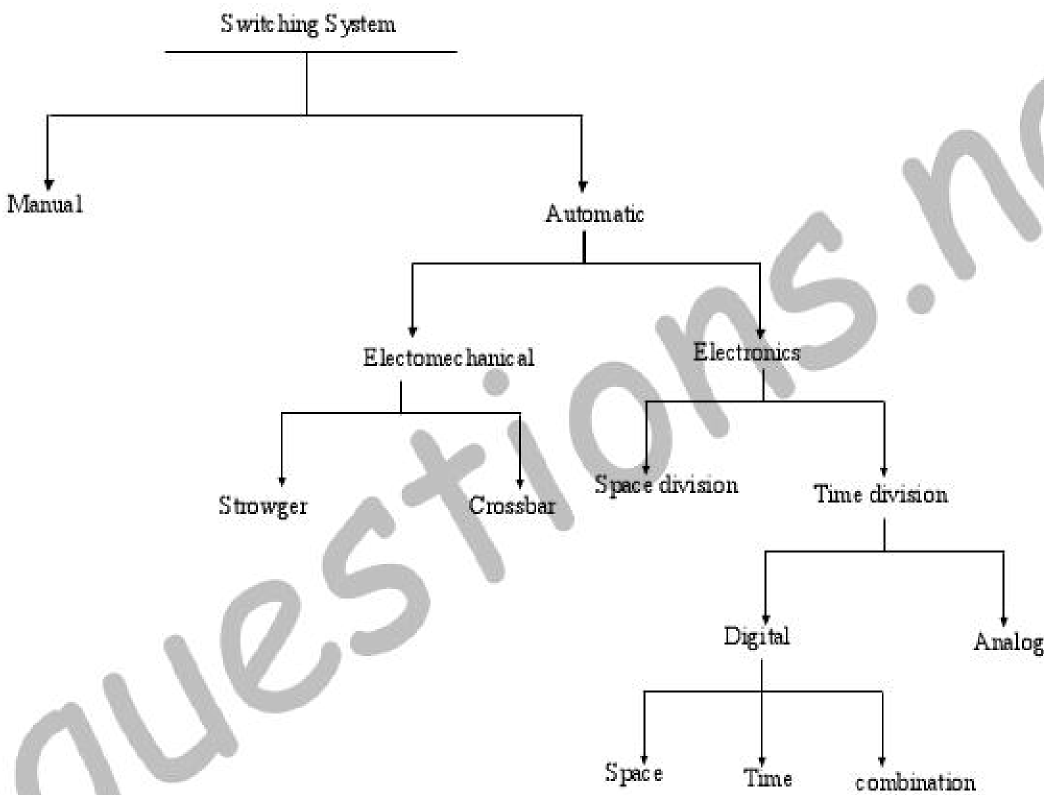
interconnection becomes more complex. To overcome this, the hierarchical model among switching system was brought to the practice.



A modern telecommunication n/w may be viewed as an aggregate of a large no of point to point electrical or optical system.

Q. How many point to point links are required in fully connected network with 5 entities.
Solution:
No of entities (n) = 5
Therefore, Total number of links = n(n-1)/ 2 = (5*4)/2 = 10

Classification of switching system:
System which is used to establish connection between entities and outlets, carry necessary function regarding establishment and release of call, signaling function etc is known as switching system.



Manual SS: In this, human being was placed in SS that help in establishing and releasing a call.

Automatic S.S: In this, operator was replaced by machine that help in signaling, establishing and releasing a call, control function.

Electromechanical: It uses the electromagnet to move the mechanical parts to perform required switching.

Strowger (Step by step): It is known as so after its inverter A.B Strowger. Control function is performed by circuits associated with switching element like uniselector, two-motion selector.

Crossbar: It consists of array of horizontal and vertical provided that horizontal and vertical contacts points are connected to these wires. The electromagnets are used to energize the horizontal and vertical crosspoints for establishment of connection between subscriber.

Electronic switching system (SPC): In this switching is possible through computer or processor.

Space division switching: In this, a dedicated path is established between the calling and the called subscriber for the entire duration of call.

Time division by switching system: In this, sampled values of signal are transmitted at fixed interval.

Digital time division s/w system: In this, binary coded data are transmitted at fixed interval.

Space SS: It is a type of digital switching where coded values are transferred during same time interval from i/p to o/p.

Time SS: It is a type of digital s/w where coded values are transferred at fixed interval of time form i/p to o/p.

Combinational: It is the combination of space and time division digital switch.

Analog time division SS:- In this, sampled voltage levels are transmitted at a fixed interval.

1. 3 Simple Telephone communication:-

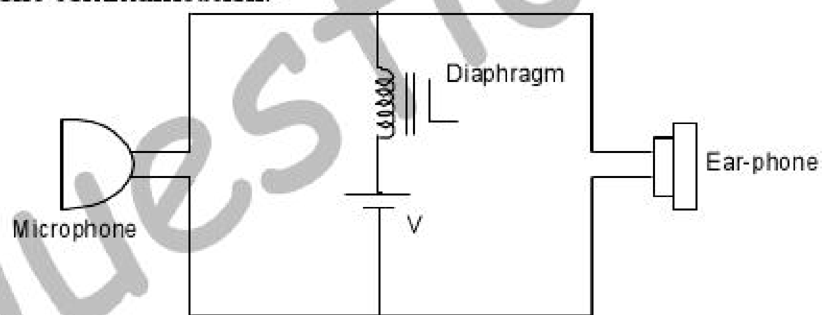


Fig: Simple comm. System.

The simplest form of telephone ckt consists of one microphone , one earphone. This ckt performs one way communication i.e simplex between two entities. The microphone and earphone acts as a transducer where microphone converts audio signal into electrical signal. Where as earphone converts electrical signal to audio signal.

Commonly used microphone is a carbon microphone which does not produce high fidelity signal but gives strong electrical signal. Here, small carbon granules is placed in a box. This carbon granules conducts electrically and the resistance offered by them is dependent upon density with which they are packed. One side of the box is mechanically attached to a diapharagm. When a sound wave impinge on the diaphragm, it vibrates the diaphragm causing the carbon particles to compress and expand, thus changing the resistivity offered by particles. If voltage is applied to the microphone, the current in the ckt varies according to the vibration of the diaphragm, when sound wave impinge on the diaphragm, the instantaneous resistance of the microphone is given by,

$r_i = r_o - r\sin wt.$

Where, r_i = instantaneous resistance.

r_o = Quiescent resistance.

r = Maximum variation in resistance offered by carbon granules. ($r < r_o$)

$r_i = r_o - r\sin wt$

$= r_o\,(1- \frac{r_i}{r_0}\sin wt)$

Now instantaneous current,

$$I = V/r_1 = \frac{V}{r_0\left(1 - \frac{r}{r_0}\sin wt\right)}$$

$$= \frac{v}{r_0}\left(1 - m\sin wt\right)^{-1} \quad \text{.........}(i)$$

Where, $m = r/r_o$, modulation factor

Now, from Binomial expansion ,

$(1\text{-m sinwt})^{-1} = (1\text{+m sinwt})$ [By neglecting higher power terms]

Equation (i) becomes,

$I = I_0 \,(1\text{+m sinwt})$ ………(ii)

Comparing this equation with amplitude modulation wave we can say that, the equation (ii) becomes the amplitude modulation. So here, microphone acts as a modulator and I_0 acts as a carrier.

Earphone usually an electromagnet with magnetic diaphragam. When electromagnet is energized by passing a current, a force is exerted on the diaphragm. The voice frequency current from microphone causes variation in the force exerted by the electromagnet, thus vibrating the diaphragm and producing sound wave the instantaneous flux linking in the poles of the electromagnet and diaphragm is given by:

$\varnothing_i = \varnothing_o + \varnothing \text{ sinwt}$.

Where,

$\varnothing_i$ = instantaneous flux linkage

$\varnothing_o$ = constant flux due to quiescent current.

$\varnothing$ = maximum amplitude of flux variable. $(\varnothing < \varnothing_o)$

The instantaneous force or diaphragm is directly proportional to the square of instantaneous flux.

i.e $F \alpha \ \phi_i^2$

$F = K\,(\varnothing_o + \varnothing \text{ sinwt})^2$

$= k\,\varnothing_o^2\,(1 + \frac{2\phi}{\phi_0}\sin wt + \left(\frac{\phi}{\phi_0}\right)^2 \sin wt)$

$\because \frac{\phi}{\phi_0} << \phi$ So highest terms are neglected.

$= k\,\varnothing_o^2\,(1 + \frac{2\phi}{\phi_0}\sin wt)$

$F = k\,\varnothing_o^2\,(1 + k_1 I_o \text{ sinwt})$

Above equation shows that force produced as diaphragm at earphone is proportional to the current produced at microphone.

Duplex:

If transmission of data takes place in both the direction, called duplex comm..

2 types: (i) Half duplex.

(ii) Full duplex.

In half duplex , transmission of data takes place in both directions but not simultaneously (e.g walkie – talkie) .
In full duplex, transmission of data takes place in both directions simultaneously. (eg. telephone system).

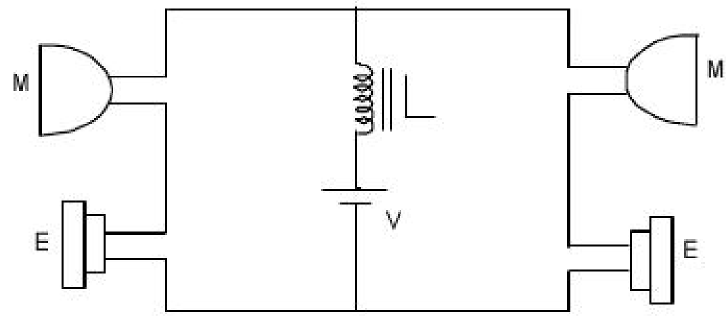
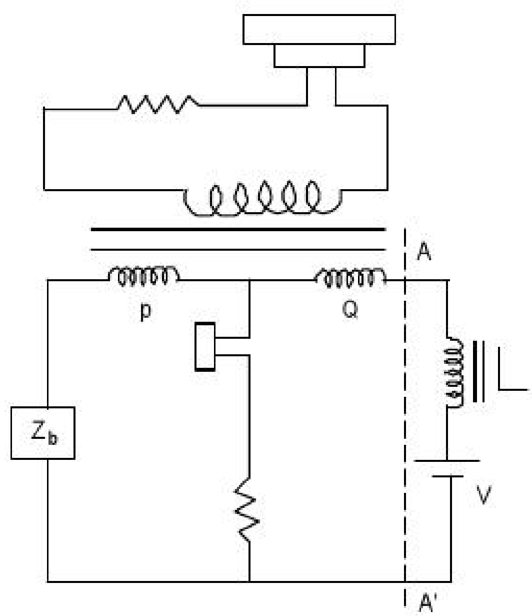


Fig: half duplex
(alternative use of microphone and earphone)
(explain as above)



Sidetone: The audio signal heard at the generating end is called side tone is known as sidetone.

Chapte:4

Basic of witching system:
Inlets /outlets : Set of input and output ckts called inlets and outlets.
Switching Matrix or networks: The hardware used to establish connection between inlets and outlets is known switching Matrix. It's provided the switching path, it is the controls subsystem of the switching system that actual establish the path.
Switching: This enable the calling subscriber to be temporary connected to call subscriber.

Switching system: It is composed of elements that peroforms switching, controlling and singnaling function.
Control function:- It's perform following task.

- Distinguish between inlets/outlets and interrupt.
- Sense the end of information transfer and release connections.
- Sense signaling information to subscriber and other exchange connected to outgoing trunk.
- Perform switching action.

Signaling: - It perform the following task.

- It gives the subscriber certains status such as dial tone busytone, ringtone, calling process etc.
- It enables to detects whether called subscriber is busy if so, indicate the same to calling subscriber.

Switching techniques:-
Figure:

Fig: Model of switching n/w

1. Symmetric switching network:-
When in above figure the N inlets subscriber is equal to the M outlets subscriber such n/w called symmetric switching n/w.

2. Folded network:-
Figur:

A n/w in which output line are folded back to the input line known as folded n/w.

3. Unfolded network:- N/w that does not support local subscriber such that the outlets are not folded back to inlets is known as unfolded networks.

Figur:

Non-folded:- With N inlets and outlets and N simultaneous information transfer are possible.
Note folded:- For N subscriber the no. of simultaneous call is equal to N/2.

 # In 100 lines f
olded n/w, how many switching elements are required for non-blocking operation.
(Ans :15)

Blocking network: The switching network that have as many simultaneous switching paths as the average number of conversion expected. In this case, it may occasionally happen that when a subscriber request a connection , there is no switching paths free in the network, and he is denied connection, in such an event, the subscriber is said to be blocked and the network is said to be blocking.
The probability that a user may get blocked is called blocking probability.

Non-blocking network:-

The switching network designed to provide N/2 simultaneous switching path, in which case the n/w is said to be non-blocking.

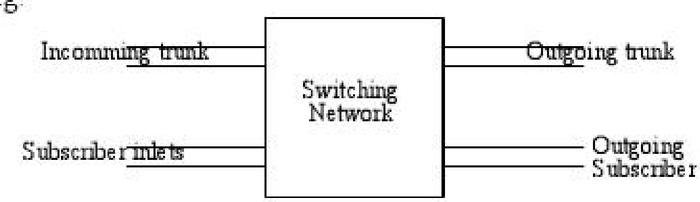


Fig: Inlet/outlet connection

From above figure , 4 types of connections may be established.

i) Local call connection between two subscriber in the system.
ii) Outgoing call connection between a subscriber and an outgoing trunk.
iii) Incomming call connections between an incoming trunks and a local subscriber.
iv) Transit call connection between an incoming trunk and outgoing trunk.

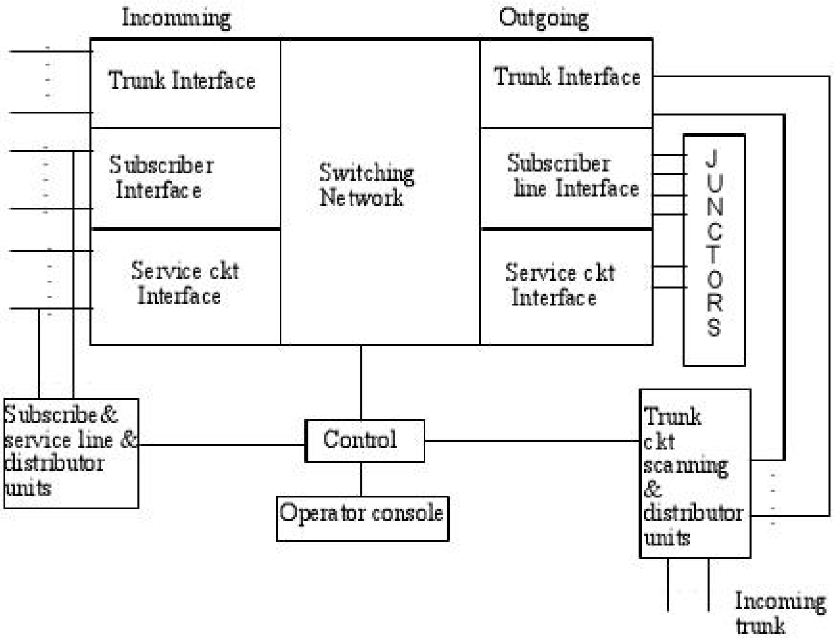


Fig: Elements of switching system.

- Subscriber lines ..... trunks are connected with subscriber and trunk line interface.
- Service ckt interfacing are used for maintenance and testing of subscriber lines.
- Junctors ckt employee a folded connection for the local subscriber and the service ckt interface but some switching system provide an internal mechanism for local connection without using the junctors ckts.
- Line scanning ckt senses and obtain signaling information from the respective lines.
- Distributors units send out signaling information on the respective lines.
- Operator console permits interaction with switching system for maintenance and administrative purpose.

Classification of switching system:

On the basis of control sub system there are manly 2 types switching system.

i) Direct control SS.

ii) Indirect control SS.

Direct control SS.

Switching system where control sub system may be an integral part of the switching matrix itself is known as direct control SS. For eg. strowger or step by step switching system.

Indirect control SS (or common control SS):

Switching system where the control sub system is outside the switching network are known as common control SS or indirect control SS. For eg. crossbar , electronic exchange.

* On the basis of generation:

1) 1st generation SS. Eg. strowger.
2) 2nd generation SS. E.g crossbar
3) 3rd generation SS. Eg SPC or electronic exchange.
4) 4th generation SS. Eg. ISDN.

Strowger SS: It was the first switching system developed by A.B strowger in 1889.

There are two types of selectors which form the building blocks for the switching system.

a) Uniselector: It is one which have a single rotary switch with a bank of contact.
b) Two-motion selector: It is capable of horizontal as well as vertical stepping movement. It has two rotary s/w.

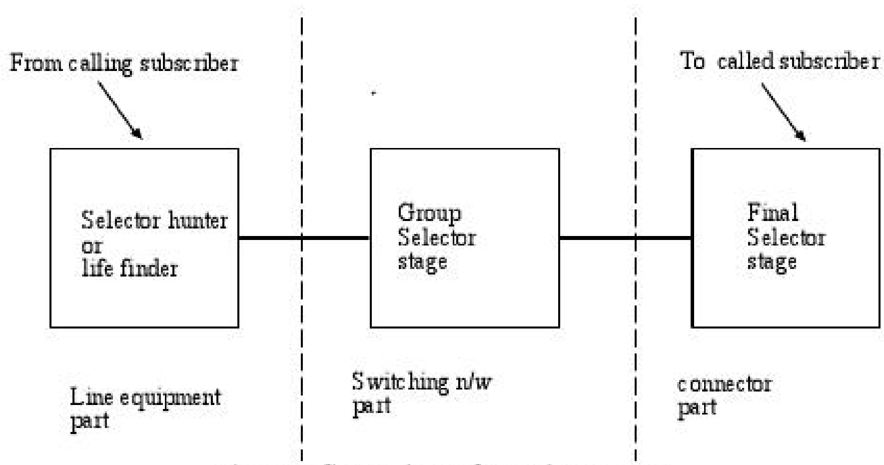


Fig: configuration of step by step ss.

It consists of 3 main parts.

1) Line equipment part.
2) Switching n/w part.

3) Connector part.

Line equipment part: - Consists of hunter or line finder. The main task of selector hunter is to search and seize a selector form the switching matrix part. There is only one selector hunter for each subscriber. Usually, 24 outlet uniselectors are used as selector hunters. Selector hunter scheme is sometime called subscriber uniselector scheme as there is dedicated uniselector for each subscriber in the system.

Line finder searches and finds the line of subscriber to be connected to the first selector associated with it. These are built using uniselector or two motion selector. Line finders and selector hunters are generally referred to as pre-selectors.

The switching matrix part consists of one or more sets of two motion selector known as first group selector, second group selector and so on.

The connector part comprises one set of two motion selector known as final selectors.

iii) Cross-bar switching:

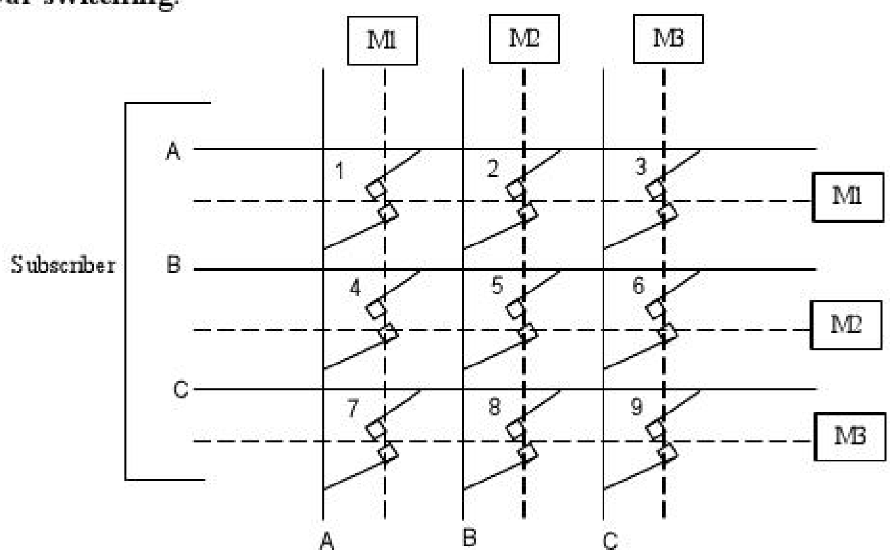


Fig: 3×3 crossbar switching

The major disadvantage of the strowger of the strowger switching system is its dependence on moving parts and contacts that are subject to wear and tear. So it was replaced by crossbar. Crossbar are designed using common control concept.

The basic idea of this switching is to provide a matrix of n×m sets of contacts with only n+m activators or less to select one of the n×m sets of contact. This form of switching is also known as co-ordinate switching as the switching contacts are arranged in a x-y plane.

When an electromagnet say in the horizontal direction is energized, the bar attached to it slightly rotates in such a way that the contacts points attached to the bars move closer to its facing contacts points but do not actually make any contact. Now if an electromagnet in the vertical direction is energized the correspondingly bar rotates causing the contact points move towards each other.

From above figure…………………….(आफै explain गर्नु)

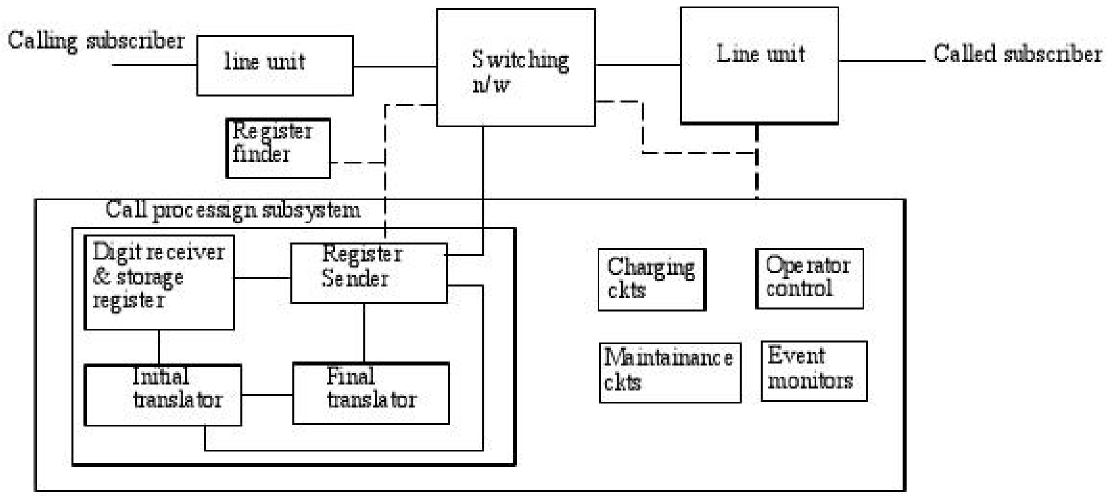


Fig: common control SS.

Where,

___ indicates data or into.

---- indicates control line.

The control function in a SS may be placed under 4 broad categories.

a) Event monitoring.
b) Call processing.
c) Charging.
d) Operation and maintenance.

Events occurring outside the exchange at the line units, trunks, junctors and signaling receiver/ sender /sender receivers are all monitored by the control sub-system. For eg. when a subscriber goes off hook, the event is sensed by the line unit, the calling location is determined and marked for dial tone.

- Register finder is activated to seize a free register.
- Initial translator determines the router for the call through the network and decides whether a call should be put through or not. It also determines the charging method and the rates on the class of service information of the subscriber which specifies details such as:

a) Call priority: When exchange or n/w is overloaded, only calls from subscriber indentified as priority call subscriber, one be put through.
b) Call baring: A subscriber may be barried from making certain calls. For eg. STD or ISD barring.
c) Call charging: It is possible to define charging rules for different subscriber in the same exchange.
d) Origin based routing: Destination of certain calls may depend on the geographical location of the calling subscriber.
e) No-dialing calls: These calls are routed to pre-determine number without the calling party having to dial.

Final translator:

It is determines the line unit to which a call must be connected and the category of the called line.

Comparison between manual exchange (SS) and automatic exchange (auto SS):

Manual: -

- The calling subscriber needs to communicate with operator in a common language.
- Privacy is not maintained
- Slow in processing.
- Time required to established and released call depends upon loads.
- Here, we cannot use electromechanical or electronic switching concept.
- Human being acts as a SS to set and release call.
- It was ancient techniques of switching.

Automatic:

- Language independent
- Privacy maintained
- -
- -
- We may use electromechanical or electronic s/w concept.
- Not required.
- Modern technique of switching.

4.2 Space division switching:- Early crossbar system were slow in processing as they used mechanical components or common control subsystem. To improve the speed of control and signaling between exchange led to the application of electronic in the design of control and signaling subsystem. Resistor and translator of the common control subsystem could be replace by a single digital computer.

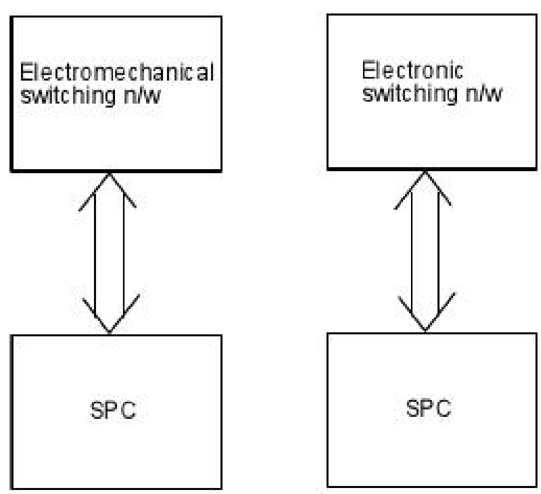


Fig: Electromechanical and electronic switching

SPC: In SPC exchange control function are carried through programmed stored in computer memory and are executed automatically one by one by the processor.

There are basically two approaches to organizing Stored program control.

1. Centralized SPC
2. Distributed SPC.

Centralize SPC: In this, all control equipment is replaced by a single processor. This configuration use more then one processor for redundancy purpose. Each processor has access to all the exchange resources like scanner and distribution points and is capable of all the control function.

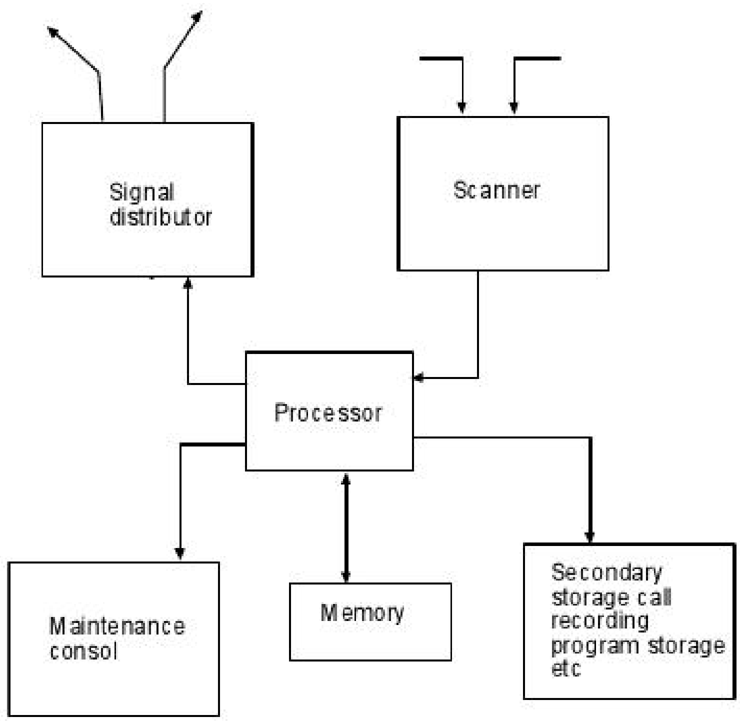


Fig: Typical centralized SPC configuration.

Present day most of electronics switching system use centralize control, only of two processor configure is used. A dual processor architecture may be configured to operate in one of three mode.

1. Standby mode.
2. Synchronous duplex mode.
3. Load sharing mode.

Standby mode:

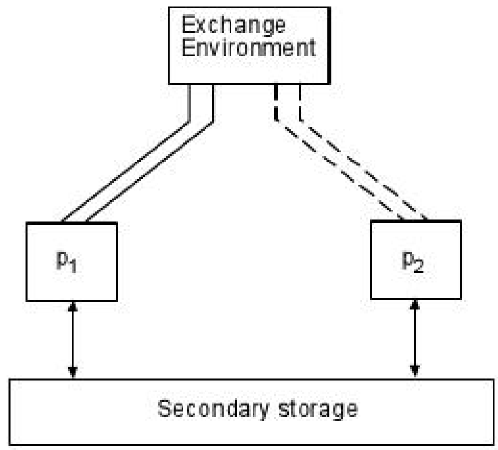


Fig: Stand by dual processor config.

- In this, one processor active and the other is on standby.
- The standby processor is brought online only when the active processor fails.

Synchronous duplex mode:-

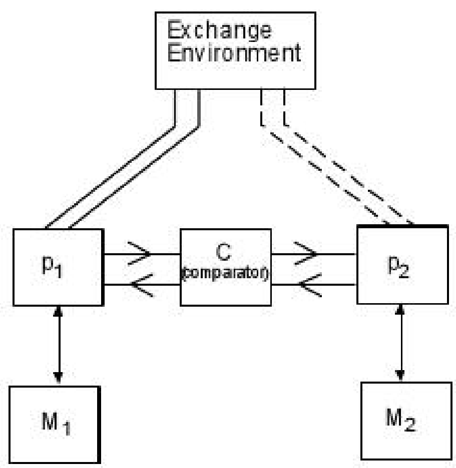


Fig: Synchronous duplex configuration.

- In this, hardware coupling is provided between the two processor which execute the same set of instruction and compare the result continuously.
- If a mismatch occurs, the faulty processor is identified and taken out of service within a few milliseconds.
- It is possible that a comparator faults occurs on account of a transient faller which does not show a when the checkout program is run. In such case, the decision as to how to continue the operation is arbitrary and three possibilities exists.

- 1. Continue with both processor.
- 2. Take out active processor and continue with the other processor.
- 3. Continue with the active processor but remove the other processor from the service.

Load shearing mode:

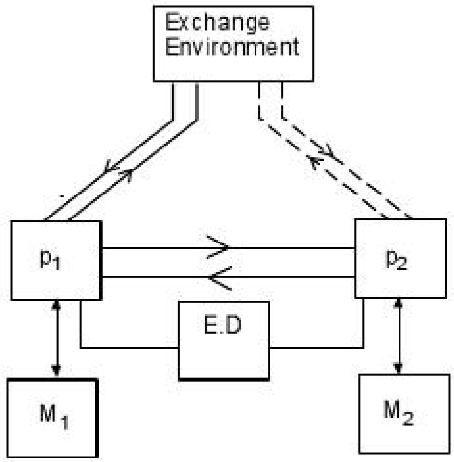


Fig: Load sharing configuration.

In this, incoming call is assigned randomly or in a predetermine order to one of the processor which then handles the call right through completion. Thus, both the processors are active simultaneously and share the load and resources dynamically.

- Exclusion device avoid to seek the same resource at the same time by both processor. This device which, when set by one the processor prohibits access particular resources by the other processor until it is ret by the first processor.
- Under normal operation each processor handles one half of the calls on statical basis.

Comparison between single and dual processor:
Availability of the single and dual processor system: -

The availability of single processor system is given by $A = \frac{MTBF}{MTBF + MTTR}$ Where,

MTBF = mean time between failure.
MTTR = Mean time to repair.

The unavailability of single processor system is given by $= 1 - A$

$$= 1 - \frac{MTBF}{MTBF + MTTR}$$

$$= \frac{MTBF + MTTR - MTBF}{MTBF + MTTR}$$

If MTBF >>>MTTR Then , $\frac{MTTR}{MTBF + MTTR}$

For dual processor, the availability is given by $A_D = \dfrac{MTBF_D}{MTBF_D + MTTR}$

Where, $MTBF_D = \dfrac{(MTBF_D)^2}{2MTTR}$

$$A_D = \frac{\dfrac{(MTBF)^2}{2MTTR}}{\dfrac{(MTBF)^2}{2MTTR} + MTTR}$$

$$= \frac{(MTBF)^2}{(MTBF)^2 + 2(MTTR)^2}$$

Unavailability of dual processor is given by,

$U_D = 1 - AD$

$$= \frac{2(MTTR)^2}{(MTBF)^2 + 2(MTTR)^2}$$

If MTBF >> MTTR then,

$U_D = 2(MTTR)^2/(MTBF)^2$

Q. Given that MTBF is equal to 2000 hrs and MTTR is equal to 4 hrs calculate the unavailability for single and dual processor system for 30 years.

MTBF = 2000 hrs

MTTR = 4 hrs.

U = MTTR/MTBF = 4/2000 = $2 * 10^{-3}$

i.e 525 hours in 30 years.

$U_D = 2\ (MTTR)^2/(MTBF)^2 = (2*16)/\ (2000 * 2000) = 8 * 10^{-6}$

i.e 2.1 hrs in 30 year.

Distributed SPC: -

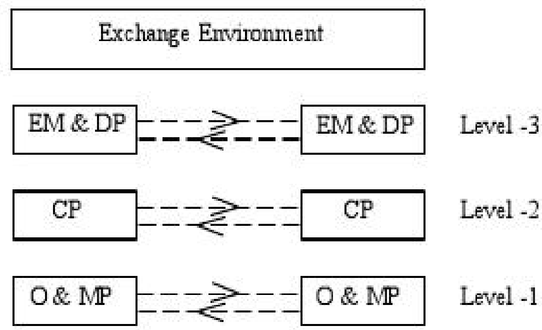
- In this, the control function are shared by many processors within the exchange itself. It uses low cost microprocessor. It has better availability and reliability then centralize SPC.
- Exchange control may decomposed either horizontally or vertically for distributed processing.
- In vertical decomposition, the exchange environment is divided into several blocks and each blocks is assign to a processor that performs all control function related to that block of equipments.
- In horizontal decomposition, each processor performs only one or some of the exchange control function.
- In this level of control functions are:

Level 3: EM & DP ……….(1)

Level 2: CP ……….(2)

Level 1: Operation and maintenance & charging ……..(3)

Where, EM & DP = Event monitoring and distribution
CP = Call processing.
Fig: Level of control function.



Level 3 processing:

- It handles scanning distribution and making function.
- Processing operation involved are of simple, specialized and well define nature.
- Processing at this level results in the setting or sensing of one or more binary condition in flip flop or register. Such simple operation are efficiently perform either by wired logic or micropropgrammed device.
- Control unit is a collection of logic circuits using logic elements, called hardwired control unit.

Level 2 processing (switching processor): -

- This processor allows data to be packed more tightly in memory so that access time will increase.
- The traffic handling capacity of control equipment is usually limited by the capacity of the switching processor. The load on switching processor is measure by its occupancy 't' , estimated by the simple formula $t = a+bN$

  Where, a = fixed overhead depending upon the exchange capacity and configuration.
  b = average time to process one call.
  N = No of calls per unit time.

Level 1 processing:

- It handles operation and maintenance function. Which involves the following steps.

a) Supports switching system hardware and software.
b) Add, modify or delete information from translation table.
c) Change subscriber class of service
d) Put a new line or trunk into operation.
e) Supervise the operation of exchange.
f) Monitor traffic.
g) Detect and locate faults and error
h) Run diagnostic and test program.
i) Man machine interaction.

Difference between microprogammed and hardware control:

Microprogamming:

1. Flexible
2. Slower
3. More expensive for moderate processing function.
4. Easier to implement complex processing function.
5. Introducing new service is easy.
6. Easier to maintain

Hardwire:
1. Not flexible
2. Faster
3. Less expensive for moderate processing function.
4. Harder to implement complex processing function
5. Introducing new service is hard.
6. Difficult to maintain

Time division switching:- Switching system, in which sampled values of speech signal are transferred at fixed interval of time. It may be analog or digital. In analog switching, the sampled voltage levels are transmitted where as in digital they are binary coded and transmitted through the system.

A time division digital switch may also be designed by using a combination of space and time switching techniques.

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Basic time division space switching:
It is generally categorized into two category.
a. i/p control time division switch.
b. o/p control time division switch.

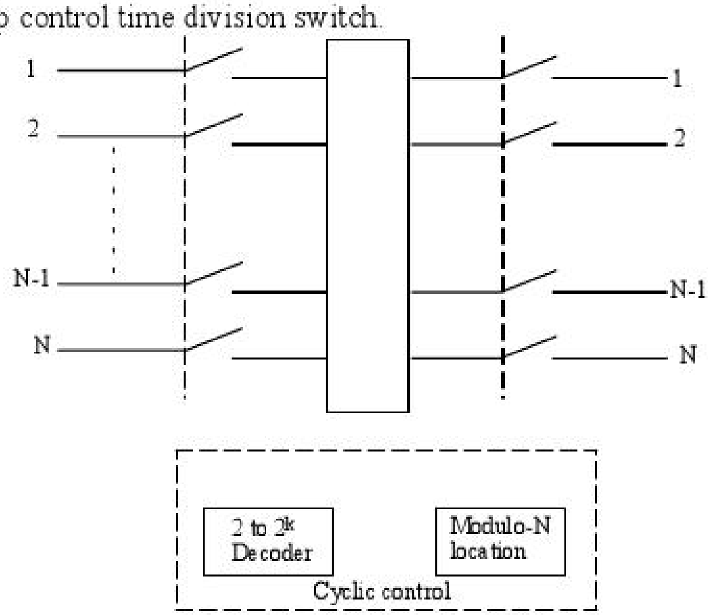


Fig 1(a) Switching structure.

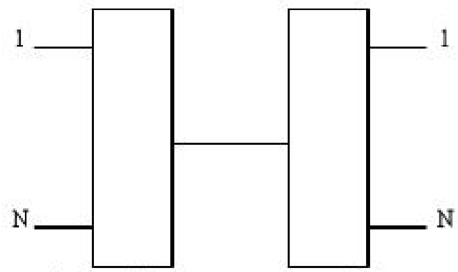


Fig:1 (b) two stage equivalent.

Fig: Simple PAM time division switching.

A simple N×N time division space switch is shown in above figure. The switch can be represented in an equivalent form as a two state network with N×1 and 1×N switching matrixes for the 1st and 2nd stage respectively. The network has only link interconnecting the stages. Each inlet and outlet is a single speech circuit corresponding to a subscriber line. The speech is carried as PAM analog samples or PCM digital samples, occurring at 125 micro second intervals.

When PAM samples are switch in a time division manner, the switching is known as analog time division switching. If PCM binary samples are switched, than switching is known as digital time division switching.

In above figure (a), the interconnecting link is shown as a bus to which a chosen inlet-outlet pair can be connected by a suitable control mechanism and speech samples are transfer from inlet to the outlet.

Time division space switching:-

It is generally categories into two groups:

a) Input control time divisions switch.
b) Output control time division switch.

Input control time division switch:-

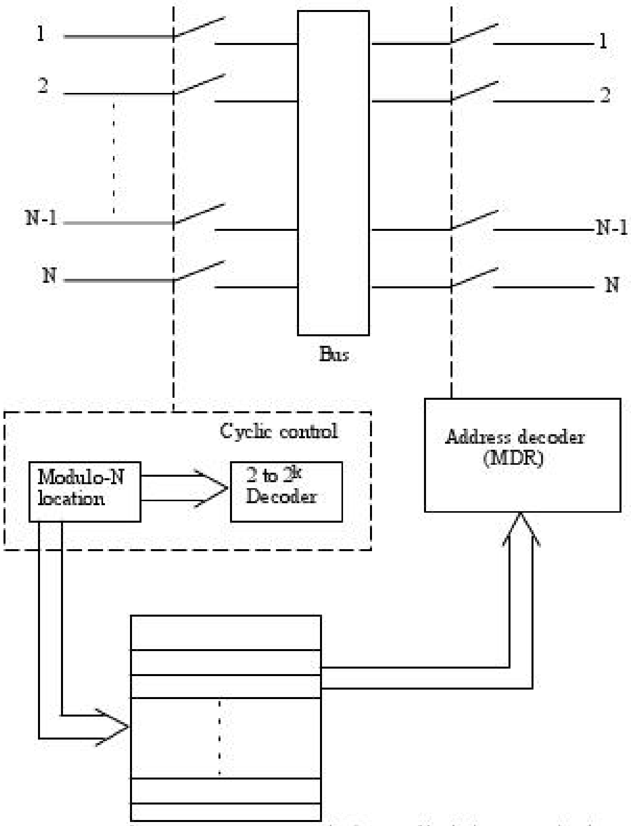


Fig: Input control time division switch.

The switch is said to be input control or input driven as the outlet is chosen depending on the inlet that is being scanned at any instant.

Modulo N counter of the cyclic control acts as the memory address register (MAR) of the control memory. Control memory has N words corresponding to N inlets and has a width of $\log_2 N$. These bits which are used to address the N outlets.

Cyclic control at the input implies that all the subscriber lines are scanned irrespective of IRRE weather they are active or not. For an active inlet i, the corresponding outlet address is contained in the i^{th} location of the control memory. It is read out an pass to the address decoder which also acts as the MDR of the control of memory.

The decoder output enables the proper outlet to be connected the bus. The sample value is than transferred from inlet to the outlet.

The bus is being shared by N connection, all of which can be active simultaneously and the physical connection is established between the inlet and outlet for the duration of sample transfer, the switching technique is known as time division space switching.

2) Output control time division space switch:

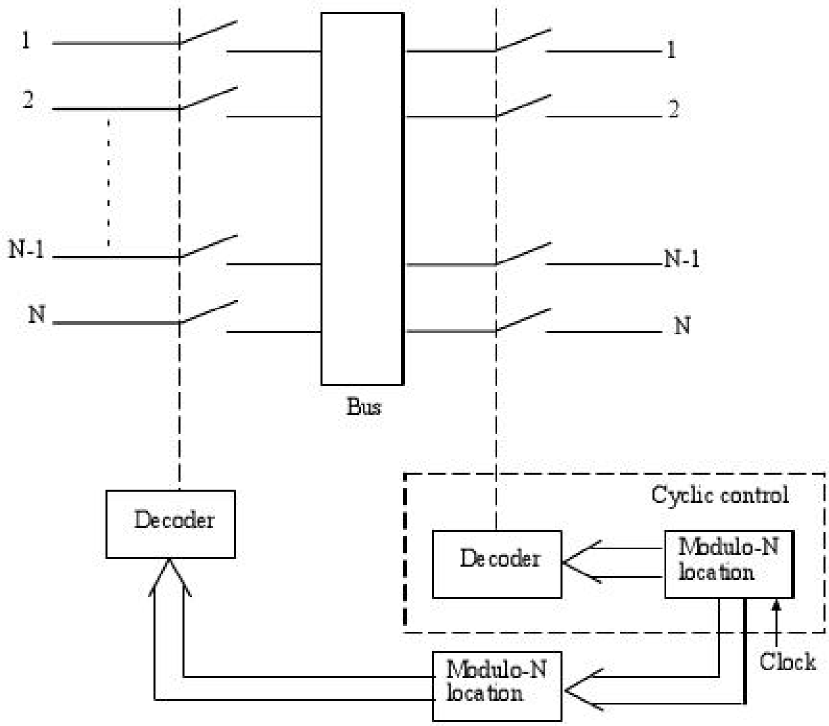


Figure: output control time division switch.

It is said to be output control because each location of the control memory is rigidly associated with a given outlet. For both input and output control configuration, the number of inlets and outlets N = switching capacity i.e

$$N = SC = \frac{125}{t_i + t_m + t_d + t_t}$$

Where, SC = switching capacity.

t_i = Time to increment the modulo-N counter.

t_m = time to read the control memory.

t_d = time to decode address and select the inlet or outlet.

t_r= time to transfer the sampled value from inlet to outlet.

The o/p controlled switches are capable of supporting broadcast connection, where i/p controlled are not.

Generalized time division space switch:-

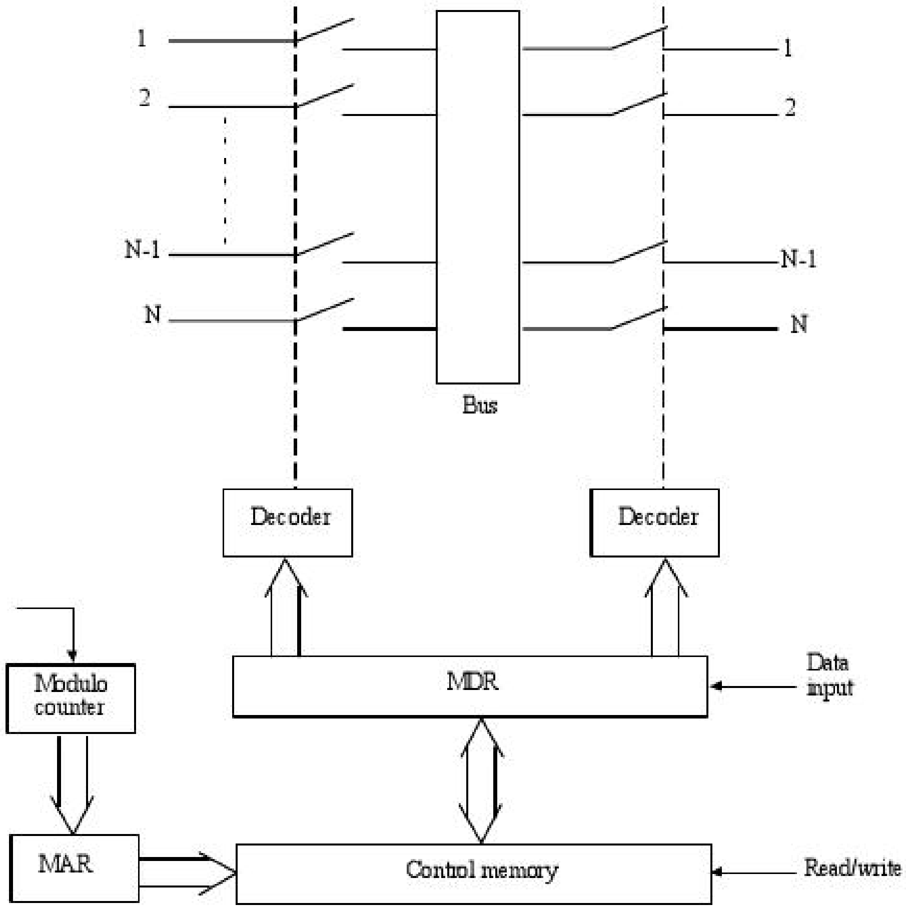


Fig: Generallised time division space switching.

Time division Time switching:-

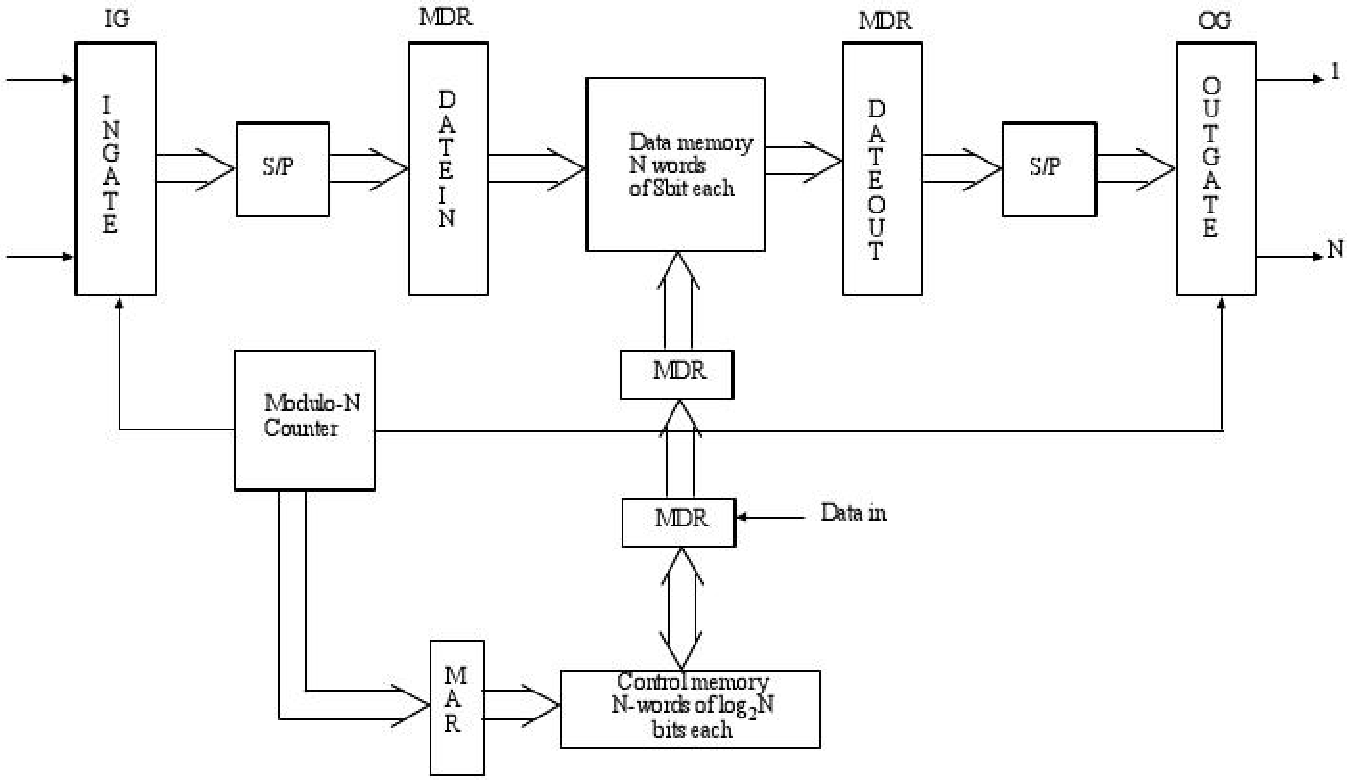
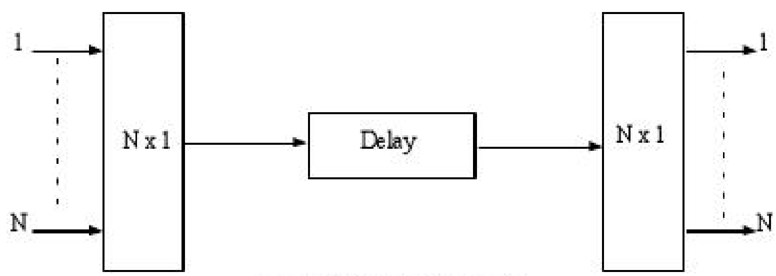


Fig: switching structure



Fig(b) Equivalent ckt

Fig- Basic time division time switching.

- In this, the data coming in through the inlets are written into the data memory and later read out to the appropriate outlets.
- The incoming and outgoing data are usually in serial form where as the data written into and read out of the memory in parallel form. Therefore necessary to perform serial to parallel conversion and parallel to serial (p/s) conversion at the inlet and outlet respectively.
- Information is not performed in real time, it is first stored in the memory and later transferred to the outlet.
- Time division time switch may be coupled in any of the following three ways:

(i) Sequential write/random read.
(ii) Random write/ sequential read.
(iii) Random input/ random output.

In the 1st two months of control, the sequential/ random read/write operation refer to the read/write operation refer to the read/write operations associated with data memory. In both cases, the inlets and outlets are scanned randomly, and the data memory is associated sequentially.

There are two modes in which time division switch may operated.

(i) Phased operation.
(ii) Slotted operation.

Phase operation:-

The phase operation of the time switches proceeds in two phase. In 1st phase, the write access to the data memory sequentially and the read access in the 2nd phase randomly.

Time taken for the two phase operation is given by

$t_s = Nt_d + N(t_d+t_c)$

Where, t_d = read/write time for the data memory.

t_c = read/write time for the control memory.

$t_d = t_c = t_m$, we have.

$t_s = 3\ N\ t_m$

Since entire operation is to be completed within 125 μs, we have expression for no of subscriber as

$N = 125/(3t_m)$ where, t_m is expressed in μs.

Slotted operation:-

In this, the 125μs period is divided into N sub-periods of duration 125/N. Each sub period i, the following operation are performed

(1) Read inlet 'i' and store the data in data memory location i.
(2) Read location 'i' of the control memory which contains the value of say j.
(3) Read the data memory location j and transfer the data to outlet i.

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3. Signal Multiplexing:

The process of sending number of separate signals together over the same transmission medium (ie metallic wire, twisted cable, coaxial cable, optical fiber cable , satellite microwave system etc) is known as signal multiplexing.

Space division multiplexing (SDM)
Frequency division multiplexing (FDM)
Time division multiplexing (TDM)

3.2 Frequency division multiplex:

The technique of separating the signals in frequency is referred to as FDM:

In this, multiple signals that originally occupied the same frequency spectrum are shifted (each) to a different frequency band and transmitted simultaneously over a single transmission medium. Thus, many relatively narrow band channels can be transmitted over a single wideband transmission system.

It is an analog scheme, the information entering an FDM system is analog and it remains analog through out transmission balanced modulator to which is fed the carrier and voice channel having frequency range of 300 hz to 3400 hz (nominal channel 0 – 4 khz).

FDM is used in telephone system, telemetry, commercial broadcast, television and communication network.

An example if the FDM signal three message signal in shown below where three band signals XYZ modulate the three separate carrier signals with the frequency F_1F_2,F_3 spaced in frequency domain with reasonable margin to avoid overlapping to avoid crosstalk & intermodulation. The output of the each modulator the added to produce a composite signal having tree multiplexed message signals.

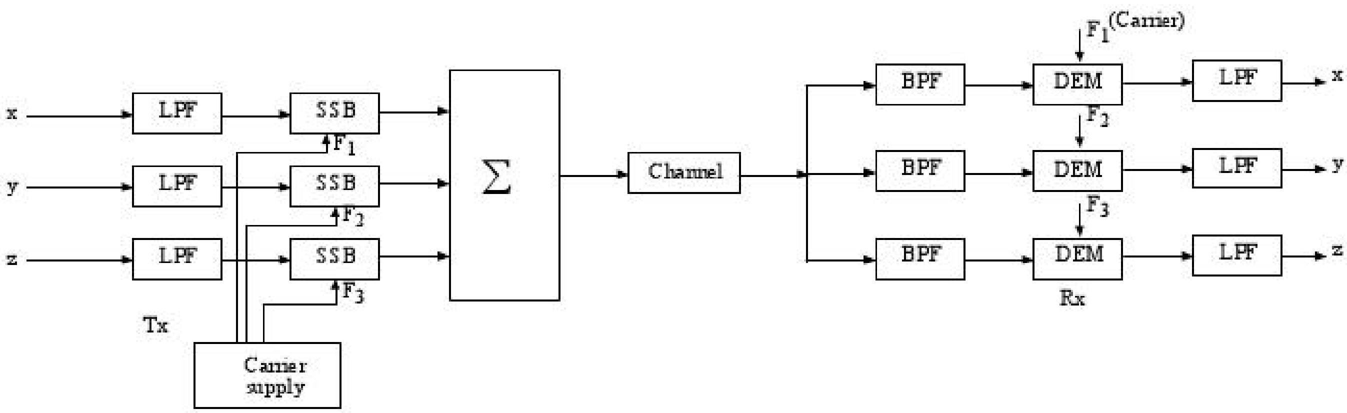


Fig: Block-diagram of FDM system.

FDM in telephony:-

Telephone channel is bandlimited to 300 – 3400 hz (BW 3100 hz) frequency slot of 4 khz is assigned to assigned to each telephone channel so that there is guard band of 900 hz for each channel. The first three channel are multiplexing at 12, 16 and 20 khz to form a pre-group of 3 telephone channels. The multiplexing (fre shifting) is SSB- USB.

FDM Hierarchy:-

1. *Message channel:-* The message channels is the basic building block of the FDM hierarchy. The basic message channel was originally intended for voice transmission that utilize voice band frequency (VB) circuit is called 3002 channel and is actually band limited to 300 hz to 3400 hz band, although for practical consideration it is considered a 4 khz channel.
2. *Basic group:* It is the first step for multiplexing the message channel. A basic group consists of 12 voice-band channels stacked next to each other in the frequency domain.
   Group BW = $12 \times 4 = 48$ khz
   $F_{out} = F_c - F_i$
   F_c = carrier (channel) frequency, F_i = channel frequency spectrum (0 – 4 khz).
   For channel 12,
   $F_c = 112 - 4\times12 = 64$ khz.
   $F_{out} = 64 - (0 \text{ to } 4 \text{ khz}) = 60 - 64$ khz.

3. *Formation of super group:-*

Five groups are combined to form a super group. The frequency spectrum for each group is 60 – 108 khz. The carrier frequency frequency for a group is derived from the following expression,

$F_c = 372 + 48n$ khz Where, n = group number.

BW being 240 khz.

4. *Basic master group:-* 5 Super group makes a master group, BW of $240 \times 5 = 1.2$ Mhz.
5. *Super master group:-* 3 basic master groups makes a super master group containing 900 telephone channels BW being 3.6 Mhz.

Filter and oscillator requirement in FDM.

In FDM the guard band between two adjacent frequency slots is not very large, the frequency stability of the oscillators should be very high in order to avoid overlapping. Generally highly stable quartz controller oscillator with stability factor of 10 – 5 or higher is employed.

Similarly SSB- filters are also necessary for the same reason mention above in oscillator.

3.3 Time division multiplexing:

The technique of separating the signals in time is referred to as time division multiplexing.

The concept of TDM is illustrated by the block diagram as shown below:

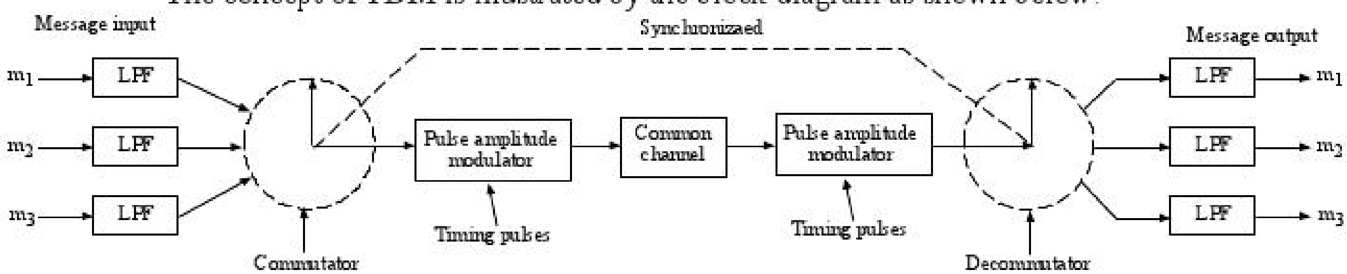


Fig: Block-diagram of TDM system.

The signals to be multiplexed are first individually band limited by low pass-filter. The low pass filter outputs are then applied to a commutator at fixed interval of time. These samples are then transmitted to the pulse amplitude modulator. The purpose of this modulator is to transform the multiplexed signal into a form suitable for transmission over the common channel.

At the receiver end of the system, the received signal is applied to a pulse demodulator, which performs the inverse operation of the pulse modulator. The narrow samples produced at the pulse demodulator output are distributed to the appropriate low-pass filters by means of a decommutator, which operates in synchronism with the commutator in the transmitter. This synchronization is essential for the satisfactory operation of the system.

The most common type of modulation used with TDM system is PCM with PCM-TDM system. Two or more voice band channels are sampled, converted to PCM codes, and then time division multiplexed onto a single metallic cable pair or optical fiber cable.

The essential operations in the transmitter of a PCM system are

(1) Sampling (2) Quantizing (3) Encoding.

The time-division multiplexed signal format is best described with reference to the Bell T_1 system, The sing signal format is given below:-

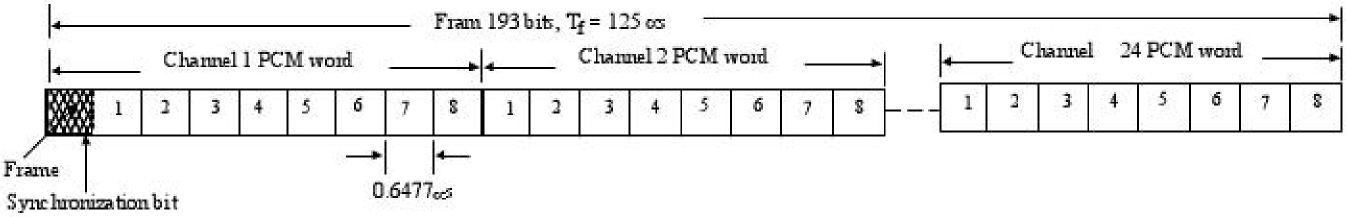


Fig: Bell T_1 PCM Format.

Each PCM word contains 8 bits and a frame contains 24 PCM channels. In addition, a periodic form synchronization signal must be transmitted and this is achieved by inserting bit from the frame synchronization codeword at the beginning of every frame. At the receiver side, a special detector termed as correlator is used to detect the frame synchronizing code word in the bit stream, which enables the frame timing to be established.

The total number of bits in a frame is $24\times8+1 = 193$ bits

From sampling theorem, sampling frequency must be at least twice the highest frequency in the spectrum of the signal being sampled. Hence, the sampling frequency for voice is 8 khz (2×4 khz) and so the interval between the PCM words for a given channel is $1/800 = 125\mu s$. Which is frame time.

Frame time(T_f): The required to transmit one sample from each channel is called frame time.

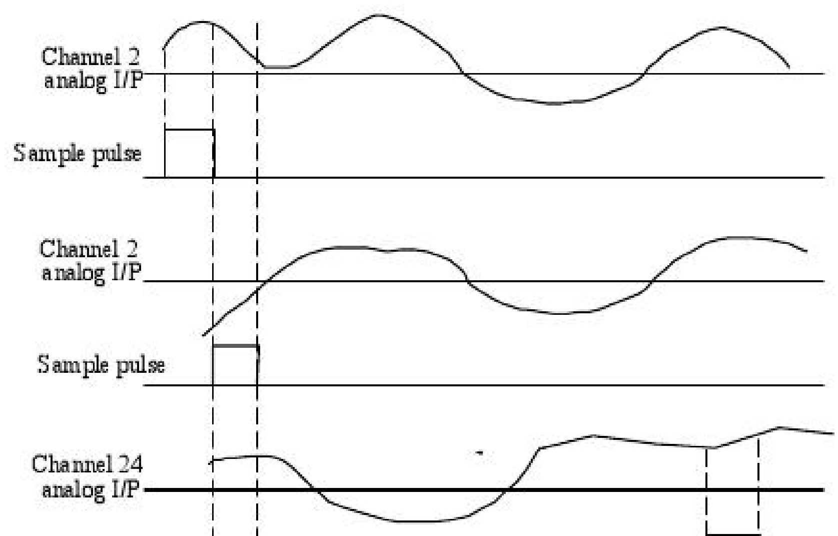


Fig: Sampling sequence.

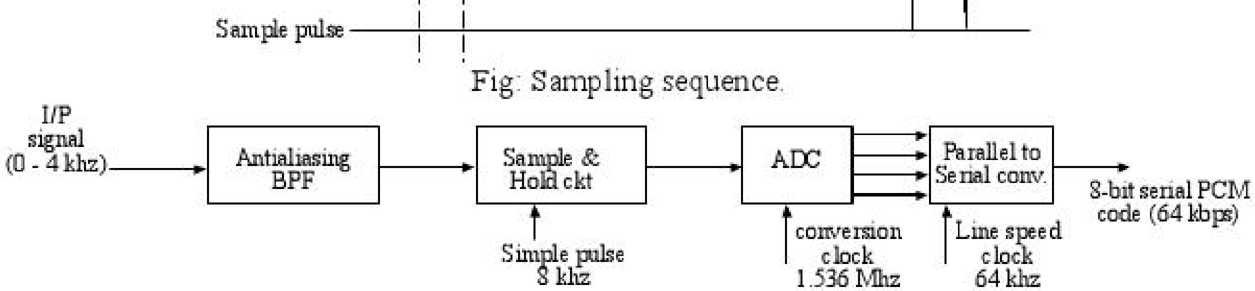


Fig: Single channel PCM transmission system.

T_1 digital carrier system:-

This system in a communication system which uses digital pulses, rather than analog signals to encode the information.

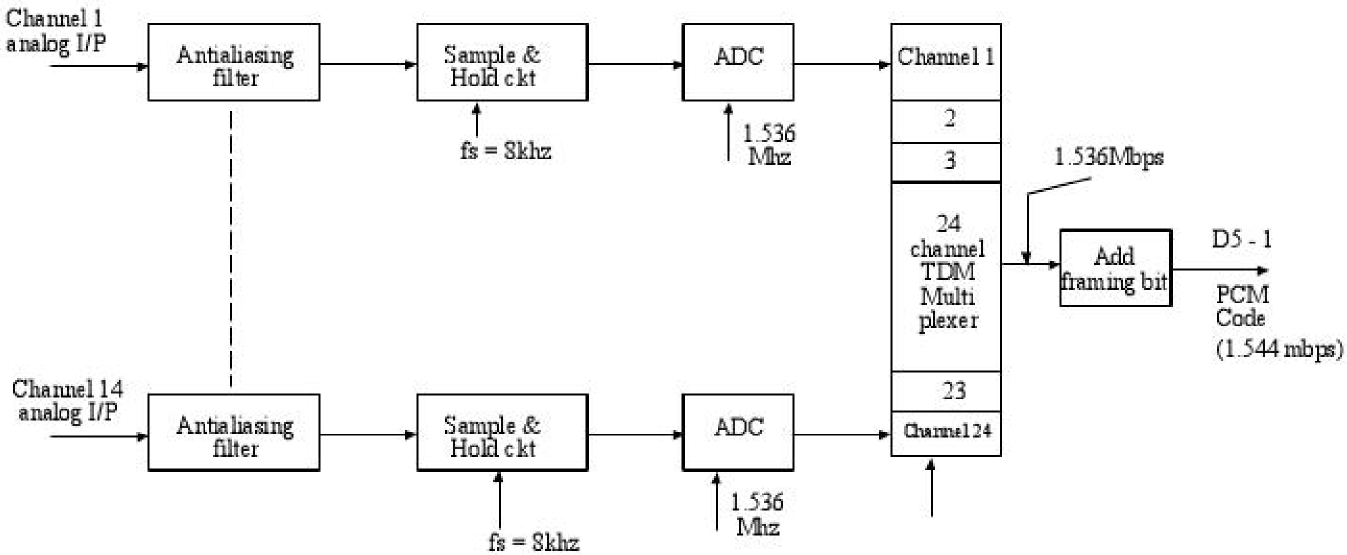


Fig: Block diagram.

3.1 Space division multiplexing(SDM):-

Technique of separating the signals in space is known as space division multiplexing. It is a rather unsophisticated form of multiplexing which simply constitutes propagating signals from different sources or different cables.

Chapter: 2

Transmission media: (TM): Physical path way that connected computer other device and people in a network is known as transmission media. Each transmission media requires specialized network hardware that has to be compatible with the medium. It also describe as the type of highway on which voice and data travel.

Characteristics of transmission media.

Each type of transmission media has special characteristics that may gives suitable for a specific type of service. Some of the important characteristics are under noted.

1. Cost: The transmission media should be cost effective.
2. Instillation requirement:-
3. Bandwidth:- It refers to the measure of capacity of medium to transmit data. High capacity means high bandwidth and low capacity means low bandwidth. It measure in BPS. Bandwidth of the cable is determined by length of the cable. A short cable accumulate greater bandwidth than long cable.
4. Band usages:-

5. Attenuation.
6. EMI (electromagnetic interference)
7. Troubleshooting.

Types of transmission media:

1. Bounded or wired or Guided
2. Unbounded or wireless and unguided.

Bounded: Twisted pair, coaxial cable, optical fiber, open wire.

Twisted pair:- A cable made of two separately insulated strands of wire twisted together is known as twisted pair. It is one of the oldest and still most common transmission media. It consists of two insulated copper wires, typically about 1mm thick. The wires are twisted together in helical form. Twisting is done because two parallel wires constitutes a fine antenna, when the wires are twisted the waves from twist cancel out, so the wire radiates less effectively.

Twisted pair can be used for transmitting either analog or digital signals. The bandwidth depends on the thickness of the wire and the distance traveled.

Twisted pair cable are of the two types:

1. UTP
2. STP

UTP(Un shield twisted pair): It consists one or more twisted pair of wires without additional shielding. It is more flexible and take less space than STP but has less bandwidth. It is suitable for both data and voice communication. It is easy to install and widely used in bus and star topologies. UTP comes in different grades called categories (Cat1 – Cat 7). Cat 1 and cat 2 are used for voice and low speed data (Telephone communication) where as 3 to 7 are used for network communication.

- It supports base band transmission.
- Minimum velocity of propagation. ($V_p = 0.59 * C$, $C = 3 * 10^8$)
- More flexible.
- Required less space than STP.
- It covers the maximum distance of 100m.
- Maximum speed supported by UTP is 10 Mbps.
- IEEE short hand for UTP is 10 BASE T.

STP: It has one or more twisted pairs within a shield.

- This shield (braided mesh or foil). This shield is used:
  1. To prevent infiltration of electromagnetic noise.
  2. To eliminate crosstalk during telephone conversation.
  3. To give protection against EMI.
  - Used for both base band broad band transmission.
  - Can be used with data rates in excess of 20 Mbps.

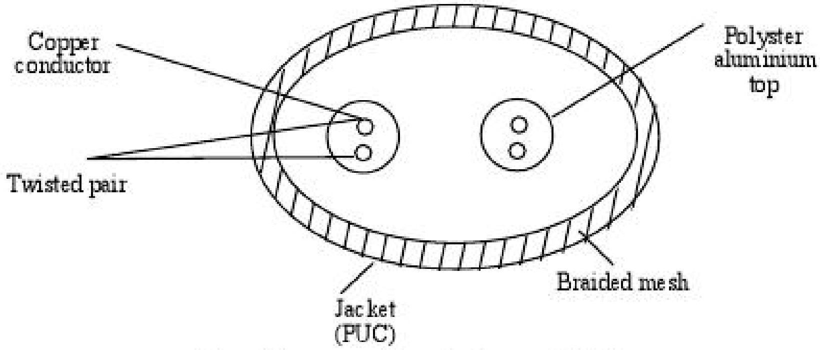
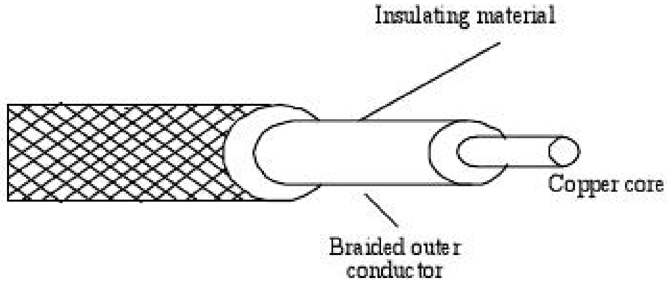


Fig: Cross sectional view of STP

Coaxial cable (Coax):-



It is common is common transmission medium widely used in television transmission. It provides higher bandwidth and better reliability than twisted. Pair it has excellent noise immunity. The bandwidth possible depends on the cable quality, length and signal to noise ratio (SNR) of data signal. It is widely used for cable TV and MAN. Modern cables have a bandwidth of close to 1 Ghz. Connector use by this cable are T, BNC , terminator etc.

There are two types of coax.

1. Thick coaxial cable.
2. Thin coaxial cable.

Thick coaxial cable:-

- Core diameter is thicker.
- Device attachment is possible at every 2.5 m.
- Maximum cable length with repeaters is 2.5 km.
- Maximum cable length is 500m.
- Data transmission speed 10 Mbps.
- Minimum velocity of propagation of velocity. ($V_p = 0.77C$)
- It support based band transmission.
- Maximum medium delay per segment.
- Uses 50 Ω terminator (when it is intended for digital transmission.)
- Short hand given by IEEE 10BASE 5.

Thin coaxial cable:-

- core diameter thinner.
- $V_p = 0.65C$

- Maximum medium delay per segment 950ns.
- Maximum length 185 meter.
- IEEE 10BASE2.

Date:2066/08/11

Microwave communication: It is widely used for long distance communication. Such as telephone communication, mobile phone, television distribution etc. It usages point to point radio transmission at frequency higher than approximately 1 gita hertz. This transmission system exists in two forms:

1. Satellite system
2. Terrestrial (earth-based) system.

Microwaves communication is characterized by the following factors.

1. The useful ranges of frequencies lies between 150 Mhz to 150 Ghz.
2. It is line of sight communication and is limited by horizon due to the curvature of the earth.
3. Signal propagation is affected by free space attenuation.
4. Frequency or phase modulation are used.

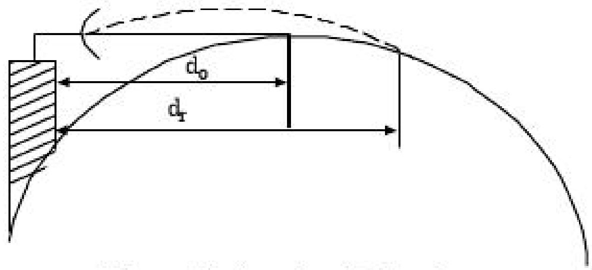
The useful frequency range of mircrowave spectrum is divided into a number of band designated by latter.

Band	Freq.range (Gigahertz)
P	0.25-0.39
L	0.39-1.55
S	1.55-3.90
C	3.90-6.30
X	6.30-10.90
K	10.90-36.00
Q	36.00-46.00
V	46.00-56.00

The bands are future divided into a number of sub bands for examples K_u bands covers the frequency range 10.9 to 14.5 Ghz in the k band.

Being a LOS limited by horizon, the height of the antenna above the earth plays the important role in determining the transmission distance in a microwave communication. Mocrowaves are usually bent or reflected beyond the optical horizon i.e horizon visible to our eyes.

The radio horizon generally future away from the optical horizon.



d_o = Optical horizon
d_r = radio horizon

Fig: optical and radio horizon.

The distance to the radio horizon varies with the atmospheric refractive changes and can be even less than the optical horizon at times. As a rule of thumb of , the optical and radio horizon is given as $d_o = 0.46\sqrt{h}$ and $d_r = 0.49h$ where h = height of the tower in m.
d_o = distance to the optical horizon in km. d_r = distance to the radio horizon in km.

Correction factor:- It is used to obtain any quantity to related to radio horizon from the corresponding quantity related to optical horizon i.e $k = d_o/d_r$. Where k is a correction factor. If k is greater than 1, than the radio horizon is nearer the optical horizon. If k < 1 the radio horizon is farther than optical horizon.

Height of the microwave tower: - Microwave tower should be such that the radio beam is not obstructed by objects like buildings, tree, mountains etc. The height must be more than the height obstacle in the way. Two factor contribute to increasing effective height.

1. Earth curvature bulge.
2. Freshel diffraction.

Earth curvature bulge:-

Earth curvature bulge is calculated as $h_{eb} = 0.078\ d_1 d_2/k$. Where h_{eb} = height increase on account of earth bulge. d_1 = distance between microwave site and obstacle in km. d_2 = distance between the other microwave site and obstacle in km.

Date: 2066/08/13

Height of microwave tower:-

Tower height should be such that the radio beam is not obstructed by objects like buildings, tree, mountains etc. The height must be more than the highest obstacle in the way. Two factor contribute to increasing the effective height.

(1) Earth's curvature bulge.
(2) Fresnel diffraction.

(1) The earth's bulge in calculated as

$h_{eb} = 0.78\ d_1 d_2$

Where, h_{eb} = height increase on account of earth's bulge.
d_1 = distance between microwave site and obstacle in km.
d_2 = distance between the other microwave site and the obstacle in km.

(2) Fresenel diffraction:- The fresenel phenomenon stuns from the fact that electromagnetic wave fronts expand as they travel through space. The expanding properties result in reflection and phase

transmission as the wave passes over obstacles. Hence, additional clearance is required above the obstacle to avoid such problem. The clearance, in required is expressed in terms of the fresenel 1st zone, 2nd zone etc. The Radius of the 1st fresnel zone R in meters is calculated as

$$R = 17.3\left[\frac{d_1 d_2}{F(d_1 + d_2)}\right]^{\frac{1}{2}}$$

Where, F = frequency of transmission n Ghz.

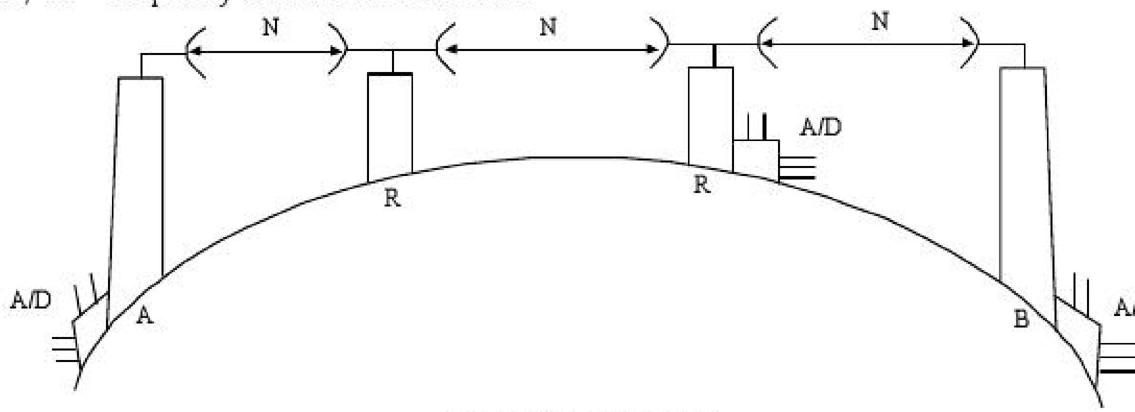


Fig: Microwave link

A, B = terminal
R = Repeater
A/D = Add and drop links

It is made up of two terminal sites and usually one or more repeater sites. A T_x site voice channel are multiplexing into baseband signal and then modulated into RF carrier- At Rx site, RF carries demodulated and the resulting baseband signal demultiplexed into individual voice channels. A repeater site is characterized by two antenna for the two directions. It receives, amplifiers and retransmit RF signal to the next site in sequence. Some repeater may add or drop of a few lines for local connection.
(a) long – haul type or (b) Short haul type.

(a) Long- haul type:- In this, only and small numbers of add or drop points but has many simple repeaters and covers a long distance end to end.

(b) Short haul type:- This system consists of a relatively small numbers of repeaters with frequent add and drop points.

Three commonly used microwave carrier band 4,6 and 11 Ghz, the 4 Ghz is used for long haul and the 6-Ghz band is useful for both long and short haul communication.

Antenna:

It required very high gain antenna.

Consider two isotropic antenna separated by a distance D. The power interrupted by receiving antenna is given by

$P_R = P_T\lambda^2/(4\pi D)^2$

Where, P_R = received power.
P_T = Transmitted power.
λ = wavelength of operation.
D = distance between two antenna.

The path loss is decibel is given by

$$L = 10\log\left(\frac{4\neq DF}{3\times 10^{8}}\right)^{2}$$

Where, D = in km and f in Ghz.

L = 20 log (4000 π/3) DF

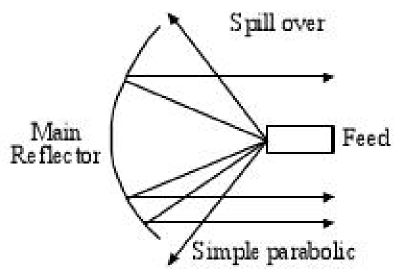
L = 92.4+20log (DF) dB.

Parabolic reflectors are used to realize highly directional high gain antenna concentrating microwave energy into parallel beam, Three versions of parabolic antenna are:

(i) Simple parabolic.

(ii) Horn.

(iii) Cassegrain.



Wave guides:-

Microwave energy is guided to the antenna feed from the transmitter system by means of waveguides. Depending upon the shape of the cross-section , the wave guides are classified as

(1) Rectangular wave guides.
(2) Circular wave guide.
(3) Elliptical waveguide.
(4) Square waveguide.

The wave guide size determines the cut-off frequency i.e a frequency below which satisfactory cannot take place. Therefore, systems operating in different band use different sizes of guides. For eg. for operation in the 3.7 – 4.24 Ghz range , a rectangular wave guide use. Wave guides are coded by a letter to indicate the shape of the cross-section and a number to indicate the larger dimension of the shape. For eg. WR-5-75 means rectangular wave guide with the width being 5-75cm.

Some of the microwave components are

1. Microwave tubes:-

(a) Multicarity klystron amplifier.

(b) Reflex klystron Oscillator.

This tube performs the function of generation and amplification in the microwave portion of the frequency spectrum. This tube consist following component.

(c) Travelling wave tube (TWT)

(d) Backward wave oscillator.

(e) Magnetrons

(f) Crossed – field amplifiers.

Klystrone: - These are velocity modulated tube that are used in radar and communication equipment as oscillators and amplifier.

2. Solid state microwave devices.

(a) Gunn diode oscillator.
(b) Imapatt diode oscillator.
(c) Trapatt diode oscillator.
(d) Masers (Microwave amplification by stimulated emission of radiation): It is same as LASER except that it works at microwave frequency
(e) Varactor diode.
(f) tunnel diode.
(g) Varactor diode.
(f) Tunnel diode.
(g) PIN diode.
(h) Mosfet
(i) MIC (Microwave integrated ckt)
(j) Microstrip line.

Applications of Microwave

It is used in different areas.

(1) Industry.

(a) Measurement of thickness of metal sheets in rolling mills.
(b) Continuous measurement of diameter of wires.
(c) Measurement and monitoring of moisture content.
(d) heating, cooking and processing of flood.

(2) Medical

(a) cancer treatment.

(3) *Navigation*

(4) *Remote sensing:-* Radar uses microwave radiation to defect the range, speech and other characterization of remote object.

(5) *Spectroscopy:-* It is the study of the interaction between radiation and matter as a function of wave length (λ).

(3) Rader and communication system.

Numerical problems:-

1. *In the path profile of a microwave link of 25km, a hill of height 70m with trees is encountered at a distance of 10km from transmitting end. Carrier frequency is 6Ghz. Determine tower height required. Assume a correction factor of 0.9 for ray bending.*

Soluiton:

Earth's bulge $(h_{eb}) = 0.078 d_1 d_2 / k$

$$= \frac{0.078 \times 10 \times 15}{0.9} = 13m$$

Fresnel diffraction $R = 17.3\left(\frac{d_1 d_2}{F(d_1 + d_2)}\right)^{\frac{1}{2}}$

$$= 17.3\left(\frac{10\times 15}{6\times 25}\right)^{\frac{1}{2}}$$

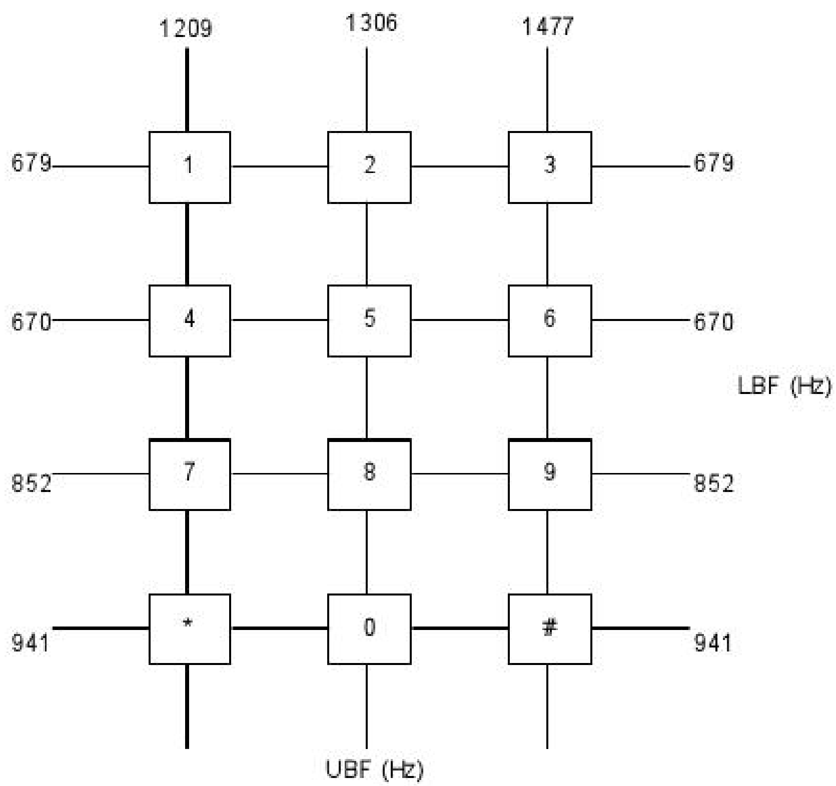
$$= 17.3(1)^{\frac{1}{2}} = 17.3$$

Allowing 15m for trees and future growth, we have
Height of the tower $= 70 + 13 + 17.3 + 15 = 122.3$

(2) See page no 340 (Example 9.6) (Vishwanathan)

Date: 2066/08/25

Touch tone dial telephone:



Structure of LAN(local area network):

- Network which spreads within small geographical area is called local area network. It is combination of computer hardware and transmission media that is relatively small.
- It is confined to a small building or group of building generally belonging to the same organization.
- It is widely used to connect PC and workstation in company offices and factories to share resources and exchange information.

  Hence, a LAN is a resources, sharing data communication with following three properties:-
  1. It is limited within the range of 0.1 – 10km.
  2. It provides high data rate (in excess of 1 Mbps).
  3. It is controlled by privative organization.

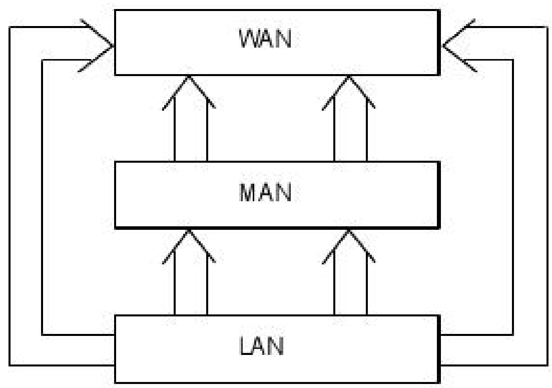


Fig : data network hierarchy

- Date networks are classified according to their geographical cover age as

1. Wide area Networking (WAN)

- Network of intercity, intercountry and intercontinental called WAN. It may be
  - (i) TDN (Terrestrial data network)
  - (ii) SBDN (Satellite based network)
  - LAN of LAN is also called WAN.

2. Metropolitan area network (MAN):

   Network which spreads within city or metropolitan is called WAN. It is generally uses community antenna television (CATV) cable and twisted pairs.

3. LAN: It may uses FON (Fiber optic network) and SONET (Synchronous optical network).

   LANs, MANs, WANs are generally interconnected in a hierarchical manner to form a global n/w (internet) as in fig (above) LANs are often directly connected to WANs.

Data transmission in PSTNs:

- Designed to carry analog voice signal.
- Can be used for data transmission.
- A modem is required (Modulator or demodulator)
- Modulator translates data pulses into voice band signals at transmitting end.
- Demodulator translates analog signals to digital information.

Data: Rates in PSTN

A voice channel in a PSTN is band limited with a nominal Bandwidth of 3.1 Khz.

Data rates in PSTN is given as

$R = 2\ H \log_2 v$ bps (Nyquist's data rate for noiseless channel).

Where bps = bits per second .

R = Maximum data rate, H = BW of the channel.

v = no of discrete levels in the signals.

$R_b = H \log_2 (1+SNR)$ (Shannol Channel Capacity theorem)

R_b = bit rate

H = B/W of the channel.

SNR = Signal to noise ratio.

Switching technique for data transmission:-

(i) Circuit Switching (Ckt switching)

(ii) Store and forward (S & F) switching.

Ckt Switching:

In this, an electrical path is established between the source and destination before any data transmission takes place.

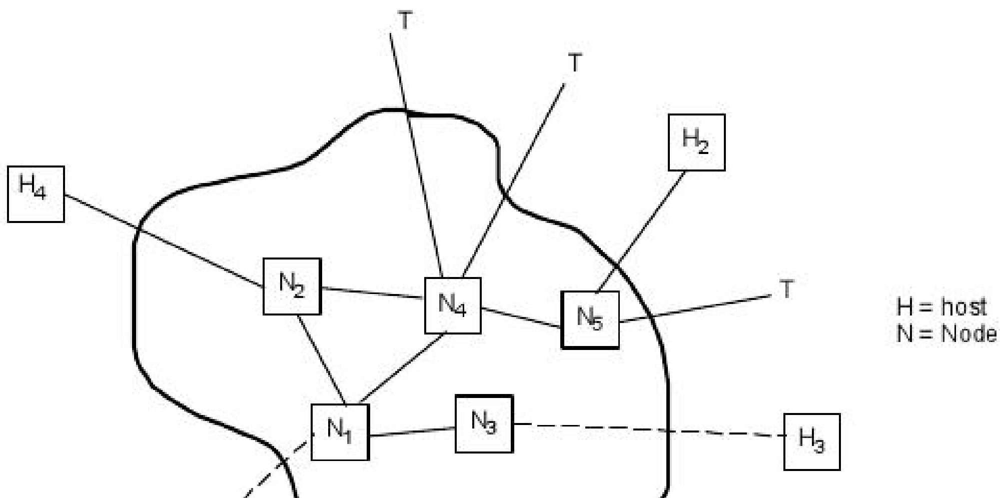


Fig: Ckt switch switched n/w

Fig shows principal of ckt switching. When the host H_1 wants to transfer data to the host computer H_3, a connection request is made to the switching node N_1 which , in turn, selects a suitable neighbouring node N_3 through which desired connection may be established.

In this switching, there are three explicit phases involved ,

(i) Connection establishment. (ii) Data transmission (iii) Connection release.

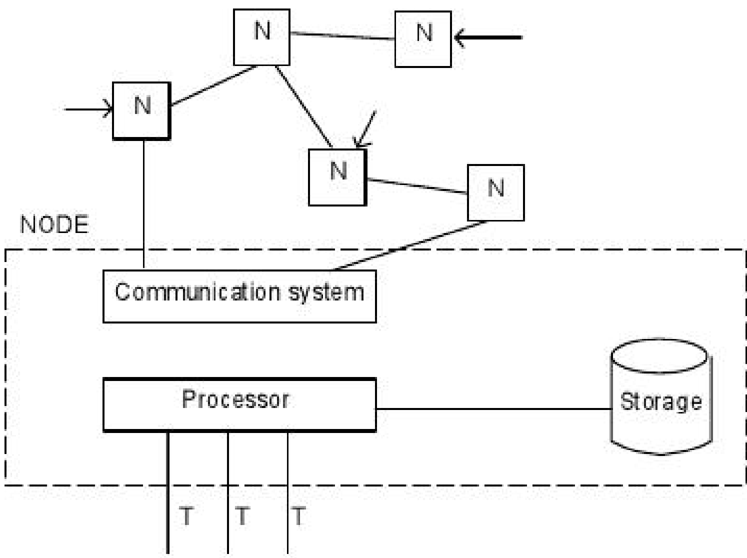
Total ckt switched time (t_{cs}) is given as

$$T_{cs} = T_e + T_t + T_r$$

T_e = Time for connection establishment.

T_t = Time for data transmission.
T_r = Time for path tramsmission.

Store and forward Switching:
S & F n/w configuration is shown in fig below:



It is classified into
(1) Message (2) Packet switching.

LAN application:

1. Office automation.
2. Factory automation.
3. Distributed computing.
4. Fire and Security System.
5. Process Control.
6. Document distribution.

Advantages of LAN:-

1. It may be put into operation with small investment, and more system may be added as the need arises.
2. It provides good back up capability.
3. It provides resources –sharing environment.
4. A LAN adhering to a certain standard permits multivendor systems to be connected to it.
5. LAN tends to exhibit an improved performance.
6. LAN offers flexibility in locating the equipment.

Disadvantages of LAN:-

1. Incremental growth makes more investment than centralized system.
2. Incapability may arise4 at the n/w , s/w and data organization level.
3. problems of security, privacy and data integrity.

LAN Technologies:-

There are three major aspect in LAN

(1) Med^{m} of transmission (2) Topology (3) Access Method.

1. **Media of transmission:**
   Twisted pair , Coax or CATV cable and Fiber optic cable.

- Twisted pair are used in low speed LANs using baseband transmission. In this mode of transmission data is to transmitted as simple electrical levels often without any modulation. Entire BW of medium is used for transmitting signals from one station.
- Coax and CATV are used for broadband transmission at speed of 10 mbps or more. It uses modulation and is suitable for transmitting data with high speed and multiplexed data.
- Fiber optic cables carry data at rates upto 100 Mbps.

2. LAN topology:

Geographical arrangement of LAN Component is known as LAN topology.

Three topology are widely used. They are

(i) Star (ii) Bus (iii) Ring.

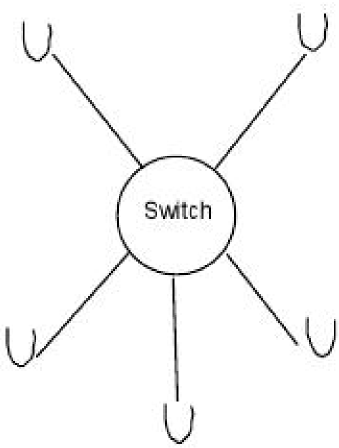


Fig (a) Star topology

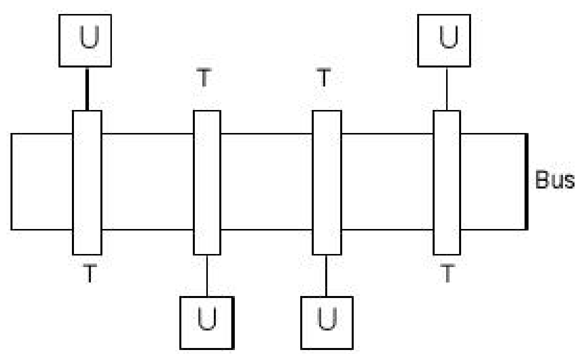


Fig (b) Bus topology.

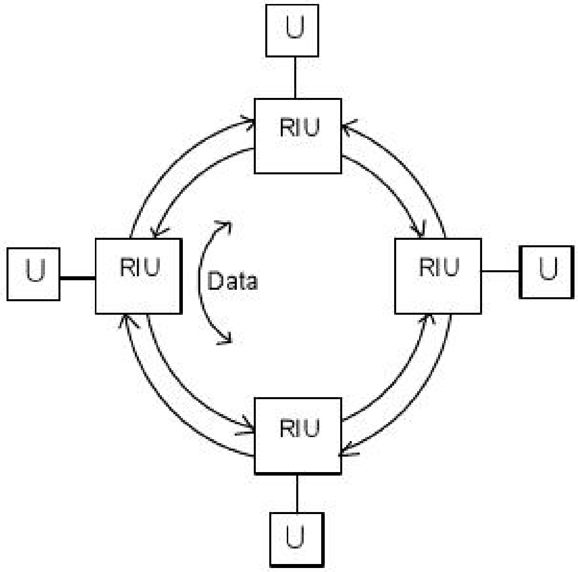


Fig (c) Ring topology.

RIU = Ring topology
U = unit (PC)
T = Terminal

Access method:

There are three access methods:

(i) Switched access Method. (Electronic switching).
(ii) Contention or multiple access method.
(iii) Token passing access method.

But all are not techniques feasible or practically possible.

Only three combinations of access techniques and topologies are popularly used.

(a) Multiple access bus.
(b) Token passing ring or Token ring.
(c) Token passing bus or token bus.

Multiple Access bus:

- Bus is broadcast medium.
- Only one data transmission can take place at any instant of time.
- It uses carrier sense multiple access (CSMA) scheme.

In this scheme, LAN station is able to know whether the channel is busy or not at any instant.

- If the bus free, transmission occurs.
- Since the channel is sensed before transmission, it is also know as (Listen-before-talk scheme).
- Performance of this bus is evaluated by throughput and maximum throughput is given as ,
- $S_{max} = \dfrac{1}{1+2B}$

- Where, B = the time expressed as a fraction of the frame time, required for all station to detect an ideal channel after transmission ends.

Another performance measure is the maximum channel utilization which is given by,

$$U_{max} = \frac{t_f}{t_f + t_p} = \frac{1}{1+a}$$

Where,

U_{max} = Maximum channel utilization.

t_f = frame time.

t_p = end to end propagation.

$a = t_p/t_f$

Between the transmission of two frames, a time gap of t_p is required.

Three variations are possible in CSMA protocols:

1. 1 – persistent.
2. Nonpersistent or zero-persistent.
3. P- persistent.

- When a station finds the channel busy, it may continue to sense the channel and transmit the frame immediately after the channel becomes free (idle). In this case CSMA protocol is known a 1 – persistent.
- The station may decide to sense the channel again after a random time when it finds the channel busy, known as zero-persistent.
- When it finds busy, the station makes a decision to persist for immediate transmission with a certain probability 'p" known as p-persistent.

Taken passing Ring LAN or Token Ring LAN:

RIU is required. RIU receive, regenerate and retransmit the data bit by bit (i.e serial transmission) on a ring. In the process RIU is capable of copying and examining every bit that phases through it. By stong group of vits received in sequence, an RIU may also examine bit pattern that travel on the ring.

Ring beign a continuous structure and the RIUs being active repeaters, data one places on the rign would go round the ring indefinitely, unless removed specifically. In contrast, data placed on a dies down automatically after shortwhile. Thus data placement and removal require consideration in a ring.

The token passing access mechanism enables a station to transmit its data on the ring. In this, one or more tokens are used to give ring access to the station.

A token is usually a 3- byte pattern as shown in fig below (a)

Fig(a): Token- Format

SD – starting delimiter byte.
AC – Access control byte.
ED – End delimiter byte.
P – Priority bits.
T – Token bit
M – monitor bit.

1	1	1	2 or 6	2 or 6		4	1	1
SD	AC	FC	DA	SA	– – – –	CRC	ED	FS

FC = Frame control
DA = Destination address
SA = Source address
CRC = Cyclic redundancy code.
FS = Frame status

Fig (b) frame format.

Fig. Token and frame formats in a token ring LAN
Steps:

1. When all stations are free, i.e ring is silent then t = 0 , free token circulates in the ring.
2. When any station has to send the data, it sets T = 1, bit and transmit the data.
3. All other station in the ring, see a busy token, examine the DA and copy the data if it is defined to them.
4. The originating station reintroduce a free token at the end of the data.
5. A tiem limt is set to hold a token by a station called token holding time (THT).
6. All the station get a chance to transmit data within a specified maximum time, known as token rotation time (TRT) which is given by,
   $TRT = N \times THT + W$
   Where, W = walk time.

Walk time is the time taken by a bit to travel around the ring and is given by,
$W = t_pL + (N/R)$.
T_p = propagation delay in $\mu s / km$
L = physical length of the ring in km.
R = Data rate in Mbps.
N/R = The delay introduced by the station in the ring with each station contributing 1 –bit delay.

7. In a ring where each RIU introduces, 1 –bit delay, the data in the ring is removed by the soruce station as shown in fig below (a) Data may also be removed by the destination station as shown in fig (b) below.

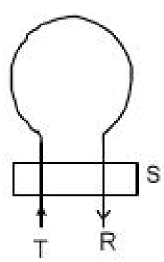


Fig (a)

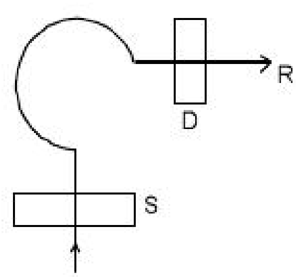


Fig (b)

Fig: Data removal in ring LANs.

T – ransmit

R – Receiver
S – source
D – destination

Token passing Bus LAN or Token bus:
It is an attempt to combine the strength of the bus architecture and the token passing access mechanism. The structure of the LAN is passive but and the stations form logical ring for token passing shown in fig below.

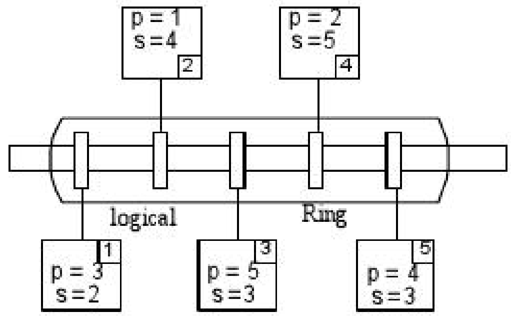


Fig: Token passing bus LAN.

Each station knows its predecessor P and its successor S. The token passes from a station to its successor. The token and frame structures are very similar to the once used in the token ring.
The strength of this LAN are

1. Robustness of the passive bus.
2. Bounded delay of token passing access.
3. No data removed consideration.
4. No constraint on the operating speed, frame size or physical length.

- In this , the logical ring is affected every time a station withdraws or joins the ring. When a station withdraws it informs its neighbours of its intention giving its P and S values which are used to change the P and S values in the successor and per decessor station respectively.
- When a station wants to join the ring, it may do so in one of two ways.

(a) It may transmit a special signal jamming the present transmission on the bus. This signal is heard by all station which then go through the cycle of establishing the logical ring. This method disrupts the operation of the bus.
(b) Alterntively, each active station periodically broadcasts a special frame to find out if there is a new station that has come up between itself and its successor. This frame is known as ‘ Solicit successor’ frame.

- When a station goes down without formally withdrawing from the ring, its predecessor recognizes this fact by observing no response on the bus when the token is passed on. It then transmits a frame known as ‘ Who follows’ frame to determine the next successor.
- Both token ring and token bus suffer from the potential danger of token being lost. A token may be corrupted by noise disturbing the token pattern which is used to recognize its presence.
- A variation of token passing bus scheme is what is known as implicit token or carrier sense multiple access/ collision avoidance (CSMA/ CA) scheme.

Networking model:

For local area network , institute of electrical and electronics engineers (IEEE) recommended that there are 7 layers of a n/w. These 7 layers network is approved by ISO (Inernational organization of standardization). This ISO is sometimes also known as ISO-OSI (Open system interconnection).

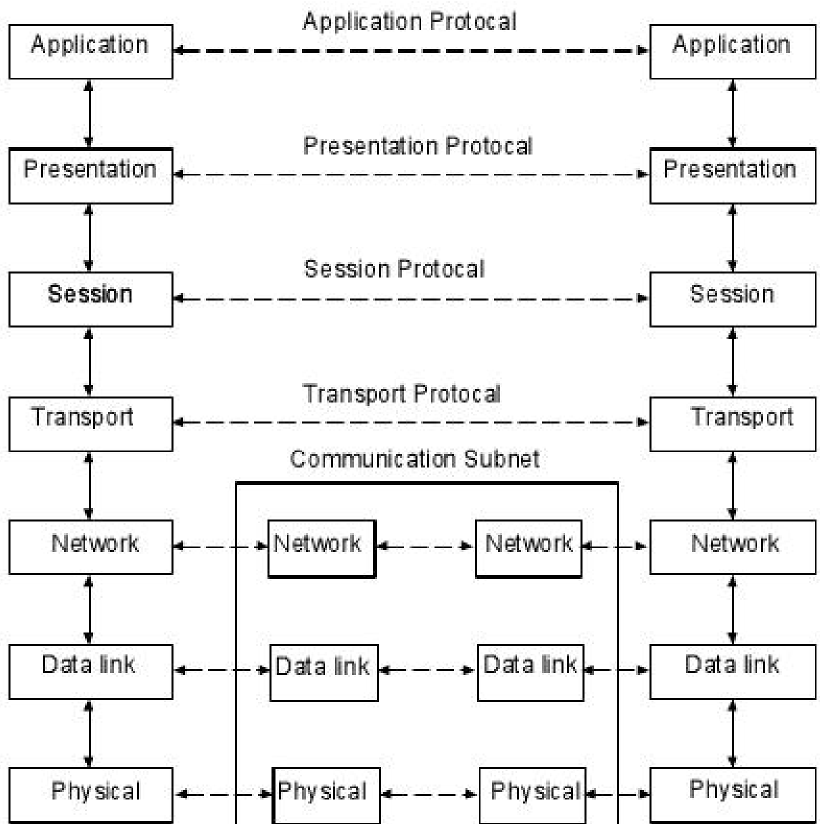


Fig: ISO-OSI reference Model

Layers: physical layers:

- It is first layer of ISO-OSI Model (ie The lowest layer of the architecture)
- This layer performs functions associated with the activation and deactivation of physical connections.
- It deals with encoding and decoding of signals.
- It transmit the data over communication link through synchronous or asynchronous medium.
- It provides mechanical, electrical , functional and procedural standard to access the physical medium.

Layer 2: Data link layers:

- This layer deals with error deflection and automatic recovery procedures required when a message is corrupted.
- It provides functional and procedural means to establish, maintain & release data link connections for the entities in the n/w layers.
- It also provides link level flow control of the frame.

Layer 3: Network layers:

- It transmits packet from the source node to destination node.
- It deals with routing and switching.
- Internetworking is an important function of the network layer.
- Network congestion is also handled by this layer.

Layer 4: Transport layer:

- It is the first end-to-end layer.
- It is responsible for matching user message characteristics and service requirements with that of the n/w capabilities.
- If performs multiplexing and splitting.
- End to end flow control and error recovery are the important function of this layer.
- When a user makes a request to the transport layer for connection, following three things may happen.

(a) Connection may be established as request.
(b) Options may be negotiated and a connection established with changed parameter values.
(c) Connection is rejected as the n/w is unable to handle even the minimal requirement of the user.

- It is based on transmission control protocol (TCP)

Layer 5: Session layer

- It organizes different session between co operating entities and perform all related functions like synchronization, failure management, control etc for the successful execution of the session.
- This layer provides the facility of activity management.

Layer 6: Presentation layer:

- It represents information to the communicating application entities in a way that preserve the meaning while resolving the differences.
- It is concerned with the representation of user or system data (ASCD or EBCDIC format).
- The standard notation used for representing information across the n/w is known as abstract syntax notation 1 (ASN.1).

Layer 7: Application layer:

- It is the highest layer in the OSI reference model.
- It provides number of services such as ,

(a) Electronic mail (E-mail) or Message handling services.
(b) Directory services.
(c) Cost allocation.
(d) Determination of quality of service (QOS).
(e) File transfer and management (FTAM)
(f) Editors and terminal support services.
(g) Telematic services like videotext.

Network interfaces:

Following equipments are widely used for n/w interface.

I. Transreceiver
II. NIC (Network interface card)

III. Transmission media Adaptor.

i. **Transceiver:** Device which is capable to transmit as well as receive data between network is known as transceiver.

- It may transmits or receiver electric, light or electromagnetic wave (EM wave).

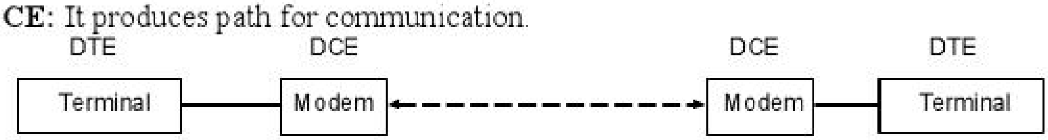
ii. **NIC:** printed ckt board which provides the connection to convert the computer's electric signals to be electric or electro magnetic signals suitable for the medium is called NIC.

iii. **Transmission media adaptor:** The NIC uses a connector that is different from what is already attached to the X'mission medium, a transmission media adaptor is used.

DTE (Data terminal equipment) and DCE (Data communication equipment) are widely used as transmission media adaptor.

DTE: Device that ends communication link is called DTE.

DCE: It produces path for communication.



Some of the important network connection are:

- RS 232C with 25 pins.
- Rj45 with 8 pins (Registered junction

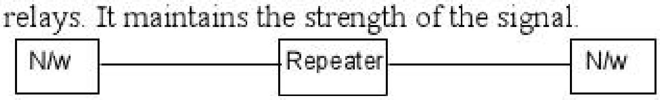
Internetworking (page 482 vishwanathan)

- Connection between network is known internetworking connectivity (i.e connection between two or more network) is possible through following devices:-

(a) Repeaters.
(b) Bridge
(c) Routers
(d) Gateways.

Repeaters:

It acts as a physical layers relays. It maintains the strength of the signal.



Bridge:

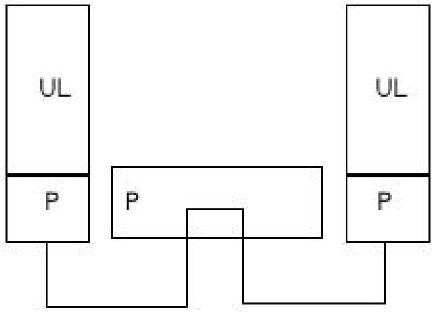
- This widely used in physical and data link layer standards. It connects two or more networks with similar protocol.

Routers:

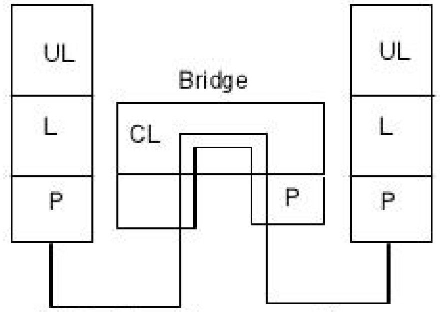
- It implements in physical, data link and network layer standards.

Gateways:

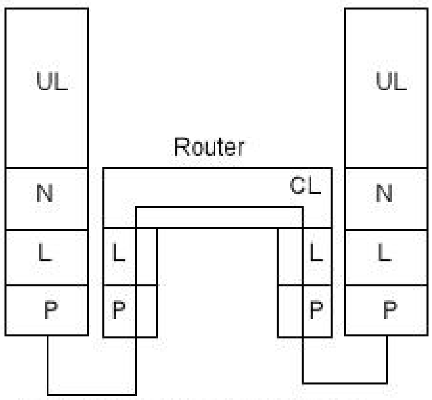
- These are considered as application relays between network environment.
- It is used to connect two or more network with different protocol.



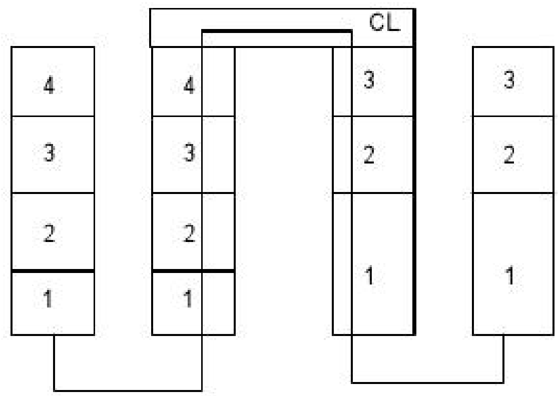
(a) Repeater interconnection



(b) Bridge interconnection



(c) Router interconnection



(d) Gateway interconnection

P – physical layer   L – data link layer   N – Network layer.
UL – upper layer.   CL – Common layer.

7. Telephone traffic and Network:

7.1 Fundamental of telephone traffic:

Engineering which provides the basis for the analysis of design of telecommunication network is known as traffic engineering. The calculation were based on quantity that specified the fraction of the time of r which a subscriber line way be busy. The task of designing cost effective networks that provide the required quantity of service under varied traffic conditions. Traffice engineering analysis enables one to determine the ability of a telecommunication n/w to carry a given traffic at a particular loss probability.

Network traffic load and parameters.

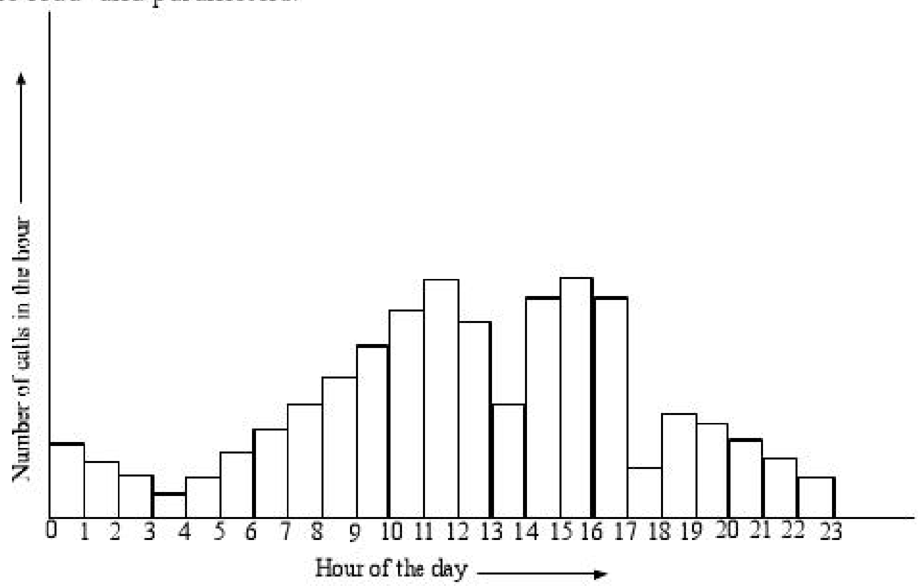


Fig: typical telephone traffic pattern on a working day.

- load carried of a communication link or channel is called traffic .
- In a day, the 60- minute interval in which the traffic is the highest is called the busy hour (BH). According to CCITT, there are three types of busy hour.

(a) Busy hour: - Continuous 1-hour period lying wholly in the time interval concerned, for which the traffic volume or number of call attempts is greatest over the days under consideration.

(b) Peak busy hour:- The busy hour each day, it usually varies from day to day, or over a number of days.

(c) Time Consistent busy hour:- The 1-hour period starting at the same time each day for which the average traffic volume or the number of call attempts in greatest over the days under consideration.

Call completion rate (CCR): It is defined as the ratio of the number of successful calls to the number of successful calls to the number of call attempts.

Busy hour call attempts (BHCA): It is the number of call attempts in busy hour.

Busy hour calling rate (BHCR): It is defined as the average number of calls originated by a subscriber during the busy hour.

Day to busy hour traffic ratio: It is the ratio of busy hour calling rate to the average calling rate for the day.

Traffic intensity: It is defined as the fraction of time for which server is busy. It is dented by A_0 and its unit is Erlang. i.e

$$A_0 = \frac{Time\ \ for\ which\ server\ is\ busy}{total\ \ period\ \ of\ \ observation}$$

It is known as carried traffic or traffic flow.
A server is said to have 1 erlang of traffic it is occupied for the entire period of observation.

A_0 is also measured in centum call second(CCS). One CCS may mean one call for 100 seconds duration or 100 calls for one second duration each or any other combination.

Sometimes, call second (CS) and call minute (CM) are also used as a measure of traffic intensity.
1E = 36CCS = 3600 CS = 60 CM. (E= eralang).

Offered traffic: Two important parameter are required to estimate the traffic intensity or the n/w load are

- Average call arrived rate (C).
- Average holding time per call (t_h).

Now , we can express the load offered to the n/w in terms of $C \uparrow t_h$ is

$A = Ct_h$ [A = offered traffic]

GOS (Grade of service)

- It is defined as the ratio of lost traffic to offered traffic. i.e

$$GOS = \frac{lost\ traffic}{Offered\ traffic} = \frac{A - A_0}{A}$$

It should be as low as possible. For e.g GOS should be in ordered of 0.002 i.e 2 calls in every 1000 calls or one call in every 500 calls may be lost.

Numericals:

Q. 1. An exchange serves 2000 subscribers, if the average BHCA is 10,000 an CCR is 60%, calculate the busy hour calling rate.
Solution:

Average busy hour calls = BHCA × CCR = 6000 calls.

$$\text{Busy hour calling rate} = \frac{average\ busy\ hour\ calls}{total\ no\ of\ subscriber}$$

Q.2. In a group of 10 servers, each is occupied for 30 minutes in an observation interval of two hours. Calculate the traffic carried by the group.
Solution:

$$\text{Traffic carried per server} = \frac{occupied\ duration}{total\ duration}$$

$$= \frac{30}{120} = 0.25E$$

Total traffic carried by the group $10 \times 0.25 = 2.5$E.

Q. A group of 20 servers carry a traffic of 10E. If the average duration of a call is 3 minutes, calculate the no of calls put through by a single server and the group as whole in a out hour period.

Solution:

Traffic per server $= \frac{10}{20} = 0.5E$

i.e server is busy for 30 minutes in one hour. No of calls put through by one server $= \frac{30}{3} = 10$ calls.

Total number of calls put through by the group $= 10 \times 20 = 200$ calls.

Q. Over a 20 minute observation interval, 40 subscriber initiate calls. Total duration of the call is 4800 seconds. Calculate the load offered to the N/W by the subscribers and the average subscriber traffic.

Solution:

Mean arrival rate (c) = 40/20 = 2 calls/minute.

Mean holding time (t_h) = 4800/ (40×60) = 2 minutes/call

Therefore, offered load = 2×2 = 4E

Average subscriber traffic = 4/40 = 0.1 E.

Modelling Switching System:-

It is possible through random process or stochastic process. Stochastic process is one in which one or more quantities vary with time in such a way that the instantaneous values of the quantities are not determinable precisely but are predictable with certain probability. The quantities are called random variables. There are four different types of stochastic processes.

i. Continuous time continuous state.
ii. Continuous time discrete state.
iii. Discrete time Continuous state.
iv. Discrete time discrete state.

A discrete state stochastic process is often called chain.

Markov process: Discrete time discrete state Markov process is defined as one which has the following property:-

$$P[\{X(t_{n+1}) = x_{n+1}\}/\{X(t_n) = x_n,\ X(t_{n-1}) = x_{n-1}, \ldots\ldots, X(t_1) = x_1\}] = P[\{X(t_{n+1}) = x_{n+1}\}/\{X(t_n) = x_n\}] \ldots (1)$$

Where, $t_1 < t_2 \ldots\ldots < t_n < t_{n+1}$ and x_i is the i^{th} discrete state space value.

Above equation states that probability that the random variable X takes on the value x_{n+1} at time step "n+1" is entirely determined by its state value in the previous time step 'n' and is independent of its state values in earlier time steps; n-1, n-2 , n-3 etc.

Birth –death process (B – D process)

B – D process are very useful in analysis of telecommunication n/w. A telecommunication n/w can be modeled as a B –D state where the number of busy servers represents the population, a cell request means a birth and a cell termination implies a death.

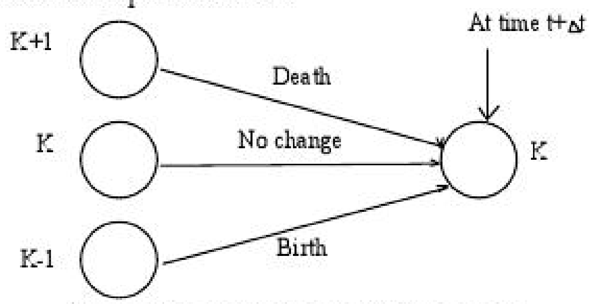


Fig: State position at a B-D process.

The B- D process moves from its state k to state k-1 if a death occurs or moves to state k+1, if birth occurs and if stays in the same state if there is no birth or death during the time period under consideration.

To analysis B-D process, we shall choose a time interval $\varnothing t$ such that,

1. There can almost be only one state transition in that interval.
2. There is only one arrival or one termination but not both in time $\varnothing t$.
3. There may be no arrival or termination leaving the state unchanged in time $\varnothing t$.

Let $P_k(t)$ be the probability that the system is in state k at time t i.e k serves are busy at time t.

λ_k = Call arrival rate in state k.

$\propto_k$ = Call termination rate in state k .

Then we have the following probabilities in the time interval $\varnothing t$.

P [exactly one arrival] = $\lambda \varnothing t$

P [exactly one termination] = $\propto \varnothing t$

P [no arrival] = $1 - \lambda \varnothing t$

P [no termination] = $1 - \propto \varnothing t$

Probability of finding the system in state k at time $t + \varnothing t$ is given by the equation,

$$P_k(t+\varnothing t) = P_{k-1}(t)\lambda_{k-1}\varnothing t + P_{k+1}(t)\propto_{k+1}\varnothing t + (1-\lambda_k\varnothing t)(1-\propto_k\varnothing t)P_k(t) \quad \ldots\ldots\ldots\ldots\ldots\ldots \quad (1)$$

Expanding equation (1) and ignoring the second order $\varnothing t$ term, we get

$$P_k(t+\varnothing t) = P_{k-1}(t)\lambda_{k-1}\varnothing t + P_{k+1}(t)\propto_{k+1}\varnothing t - (\lambda_k + \propto_k)P_k(t)\varnothing t P_k(t) \quad \ldots\ldots\ldots\ldots \quad (2)$$

Rearranging term in equation (2) we get,

$$\frac{P_k(t+\varnothing t) - P_k(t)}{\varnothing t} = P_{k-1}(t)\lambda_{k-1} + P_{k+1}(t)\propto_{k+1} - (\lambda_k - \propto_k)P_k(t) \quad \ldots\ldots\ldots\ldots \quad (3)$$

In the limit $\varnothing t \to 0$, we get

$$\frac{dP_k(t)}{dt} = P_{k-1}(t) \quad \lambda_{k-1} + P_{k+1}(t)\propto_{k+1} - (\lambda_k + \propto_k)P_k(t) \quad \ldots\ldots\ldots\ldots \quad (4)$$

Equation (4) in differential equation governing the dynamics of a B – D process equation (4) applies for all values of k ≥ 1, for k = 0, i.e no call in progress, there can be no termination of call i.e $\propto = 0$. Hence equation becomes for k =0.

$$\frac{dP_0(t)}{dt} = P_1(t)\propto_1 - \lambda_0 P_0(t) \quad \ldots\ldots\ldots\ldots\ldots \quad (5)$$

Under steady state condition, the state probabilities reach equilibrium value do not change with time i.e

$$P_k(t_1) = P_k(t_2) = P_k(t_i) = \frac{dP_k(t)}{dt} = 0$$

And the B-D process becomes stationary. Therefore, the steady state equation of a B-D process are

$$P_{k-1}\lambda_{k-1} + P_{k+1}\propto_{k+1} - (\lambda_k - \propto_k)P_k = 0 \quad for \quad k \geq 1 \quad \ldots\ldots\ldots\ldots \quad (6)$$

$$P_1\propto_1 - \lambda_0 P_0 = 0 \quad \text{for } k = 0 \quad \ldots\ldots\ldots\ldots\ldots\ldots \quad (7)$$

It may be noted that the steady state behaviour of a telecommunication switching system is governed by equation (6) and equation (7).

When the system is modeled as a B-D process.

In a system modeled as a B-D process, the termination phenomenon can be characterized as pure death process. We obtain pure death process from a B-D process by setting the birth rate equal to zero.

Blocking models and estimates:- Telecommunication system may be classified as loss system or delay system. The behavior of loss system is studied by using blocking models and that of the delay system by using quicking models. We concerned with three aspects while dealing with analysis of the telecommunication system.

(i) Modelling the system.
(ii) Traffic arrived model.
(iii) Service time distribution.

There are three models of loss system:-

(i) lost call cleared (LCC).
(ii) Lost calls returned (LCR)
(iii) Lost calls held (LCH)

Lost calls cleared system with infinite sources was first studied by A.K Erlang to estimate the blocking probability and the GOS.

Offered traffic (A) is expressed as,

A = λt_h .....(1) where, λ = average passion call arrival rate.

When all serves are busy no traffic is accepted by the n/w such a traffic on the n/w is known as Erlang traffic. In this case, we have,

$C_i = \lambda$ for $0 " i < R$, $C_R = 0$ (C = average call arrival rate).

Where, R is the no of serves in the system. The mean effective traffic C_o is calculated as

$$C_0 = \sum_{i=0}^{R-1} \lambda P_i \quad \text{Where, } P_i = \text{probability that the system is in state i.}$$

The system can be in any one of the 0 , 1, 2, 3, ...R states. Therefore we have,

$$P_0 + P_1 + P_2 + \ldots\ldots\ldots\ldots P_R = 1 \quad \ldots\ldots\ldots\ldots\ldots\ldots(2)$$

Now,

$$C_0 = \lambda(P_0 + P_1 + \ldots\ldots\ldots P_{R-1}) = \lambda(1 - P_R) \quad \ldots\ldots\ldots\ldots\ldots \quad (3)$$

The mean traffic carried by the n/w is given by,

$$A_0 = C_0 t_h \quad \ldots\ldots\ldots\ldots\ldots (4)$$

$$= \lambda(1 - P_R)t_h \quad \ldots\ldots\ldots\ldots (5)$$

From equation (i)

$$A_0 = \frac{A}{t_h}(1 - P_R)t_h \qquad (\because \quad A = \lambda t_h)$$

$$P_R = \frac{A - A_0}{A}$$ The blocking probability P_B is the same as the probability that all the server are busy i.e P_R.

Therefore, GOS $= P_B$.

Consider termination rate is directly proportional to the number of busy server as given by

$$\propto_k = k \propto for\ 0 " k " R$$

Where, $\propto$ = mean call termination rate $= \frac{1}{t_n}$

$\propto_k =$ call termination rate in state k.

We have equation for B-D process.

$$P_{k-1}\lambda + P_{k+1}\propto(k+1) - (\lambda_k - \propto_k)P_k = 0$$

Now,

$$P_{k-1}\lambda + P_{k+1}\propto(k+1) - (\lambda_k - k\propto)P_k = 0 \qquad \because (\propto_k = k\propto)$$

Equation (1), we get,

$$P_{k+1} = \frac{-AP_{k-1} + AP_k + kP_k}{k+1} \quad \text{for} \quad k > 0 \quad \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots (6)$$

$$P_1 = AP_0 \quad \text{for} \quad k = 0 \quad \ldots\ldots\ldots\ldots\ldots (7)$$

k = 1, we have

$$P_2 = \frac{AP_1 + P_1 - AP_1}{2}$$

Substituting for P_1 form equation (7) .

$$P_2 = \frac{A^2 P_0}{2}$$

Fro k = 2, we have

$$P_3 = \frac{AP_2 + 2P_2 - AP_1}{3} = \frac{A^3 P_0}{3 \times 2} = \frac{A^2 P_0}{3!}$$

Generalizing we get,

$$P_j = \frac{A^j P_0}{j!} \quad \ldots\ldots\ldots\ldots\ldots (8)$$

From equation (2) and (8)

$$P_0 + AP_0 + \ldots\ldots\ldots\ldots + \frac{A^R P_0}{R!} = 1$$

For , $P_0 = \dfrac{1}{1 + A + \dfrac{A^2}{2!} + \ldots\ldots + \dfrac{A^R}{R!}}$ ………………… (9)

For k = R , Substituting for P_0 form equation (9) into equation (8) we get,

Above formula is Erlang B-formula or loss formula.

Quenign theory:-

Delay system are analyzed using this theory which sometime known as waiting line theory.

Elements of quening system.

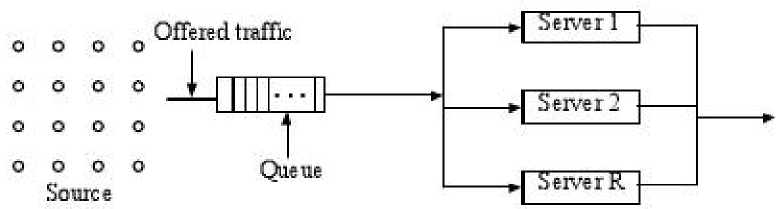


Fig: elements of queuing system.

A quening system is characterized by a set of six parameters. The notation read as A/ B/C/K/m/z. The parameter specification are as follows.

A = arrival process specification.
B = Service time distribution.
C = Number of server.
K = quene capacity.
m = number of source (Input population).
Z = Service discipline.

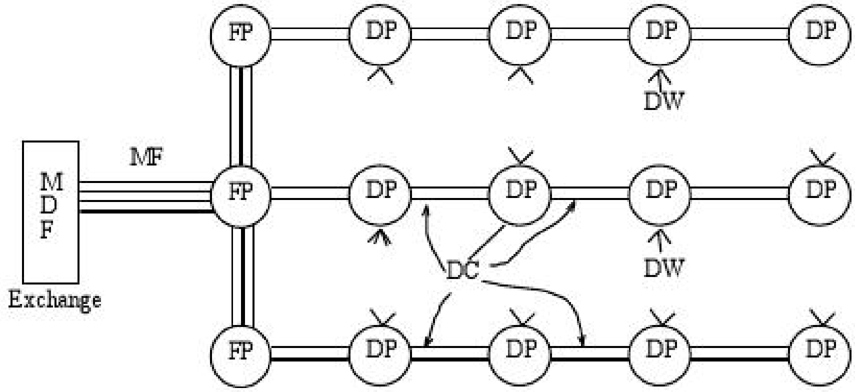
Telephone Network

- PSTN or the old telephone system (POTS) is most popular telecommunication n/w
- There are over 400 million telephone connection and over 60,000 telephone exchanges the world over.
- The length of telephone exceeds a billion km.

A telecommunication n/w may be viewed as consisting of the following major system.

(1) Subscriber end instruments or equipment.
(2) Subscriber loop systems.
(3) Switching systems.
(4) Transmission system.
(5) Signaling systems.

Subscriber loop system.



MDF - Main distribution frame, FM – Main feeder.
FP - Feeder point BF – branch feeder.
DP – Distribution point DC – distribution cable.
DW - Drop wires.

Fig: Cable hierarchy for subscriber loop.

Every subscriber in a telephone n/w is connected generally to the nearest switching office by means of dedicated pair of wires.

Subscriber pairs and exchange pairs are interconnected at the MDF by means of jumpers. The MDF thus provides a flexible interconnection mechanism which is very useful in reallocating cable pairs and subscriber numbers.

It is desirable form economy point view that the subscriber loop lengths are as large as possible so that a single exchange can serve a large area. But two factors limit their length.

(1) Signaling limits.
(2) Attenuation limits.

ISDN (Integrated service digital network) An integrated digital network in which the same digital switches and digital paths are used to establish different services, for example telephony and data.

ISDN based on six conceptual principles:

1. ISDN will be based on and will evolve from the telephony IDN .
2. Now services introduced into the ISDN should be so arranged as to compatible with 64kbps switched digital connection.
3. The transition from the existing network to a ISDN may required a period of time (extending over one or two decades).
4. During the transition period, arrangement must be made for the interworking of services on ISDNs and services on other n/ws.
5. The ISDN will content intelligence for the purpose of providing services features, maintenance and n/w management function.

6. A layered functional set of protocols appears desirable for the various access arrangements to ISDN.

Motivation for ISDN (Three factors are responsible for the development towards the ISDN)

(2) Sociological or Societal needs.
(3) Economic necessity.
(4) Technology developments.

New services:

(1) Videotex.
(2) E-mail
(3) Digital Fax.
(4) Teletex.
(5) Database access.
(6) Electronic fund transfer.
(7) Image and graphics image.
(8) Document storage and transfer.
(9) Atomatic Alarm services eq. smoke, fire, police and medical.
(10) Audio and video conferencing.

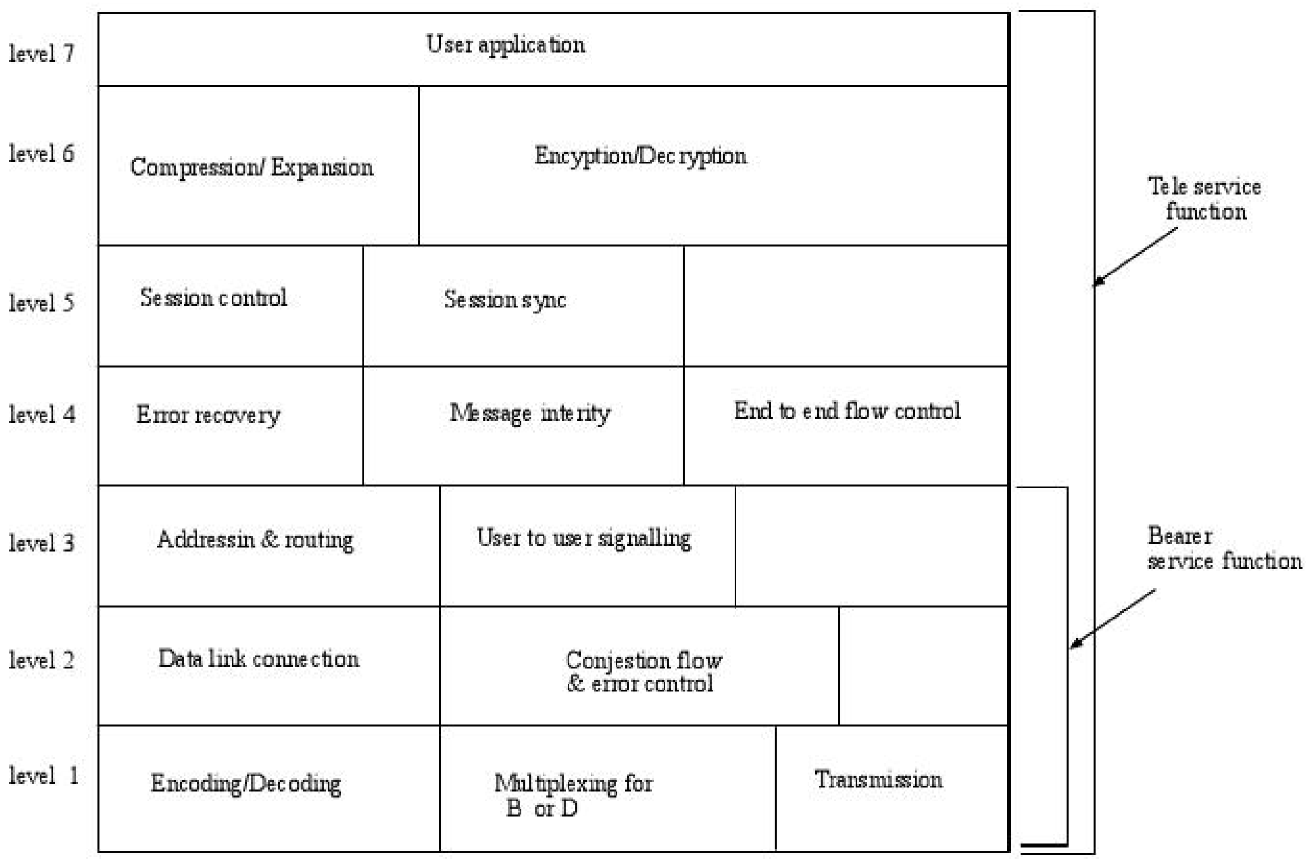


Fig: ISDN protocol architecture.

Transmission channel: -

There are three types of fundamental channels in ISDN. These are:

1. Basic information channel B channel, 64 kbps.
2. Signalling channel D channel , 16 or 64 kbps.
3. High speed channel H channels.

H_0 channel , 384 kbps, H_7 channel 1536 kbps, H_{12} channel, 1920 kbps.

Signaling: (Signalling in ISDN falls into two distinct categories)

1. User level signaling 2. Network level signaling.

ISDN service categories:

ISDN services

1. Barrier services
   a. Basic bearer services
   b. Basic teleservice + supplementary services.
2. Teleservices.
   a. Basic teleservices.
   b. Basic barrier service + supplementary services.

Numerical: (see page 311 vishwanathan)

Q. A group of 20 services carry a traffic of 10 erlangs. If the average duration of call is 3 min. Calculate the numbers of calls putthrough by a single server and the group as a whole in a 1 hrs period.

Solution:

$A_0 = 10E$

Occupied time = 3 min

Total server = 20

Traffic per server = 10/20 = 0.5E

Time occupied for 1 server = 0.5 = t_o/t

$\therefore t_0 = 0.5 \times 60 = 30$ min

Number of calls made by single server (put through) = 30/3 = 10 calls.

The number of call put through by total server = $10 \times 20 = 200$.

Relation:

1 E = 36 CCS

= 3600CS

= 60Cm

CCS = Centum call second.

Cs = call second.

Cm = call minutes.

Q. A subscriber makes three phone calls of 3 minutes, 4 minutes and 2 minutes duration in 1 hrs duration. Calculate the subscriber traffic in erlangs, CCs and Cm.

$$A_0 = \frac{4+3+2}{60} = 0.15E$$

$$A_0 = 0.15 \times 36ccs$$

$$= 5.4ccs$$

$$= 5.4 \times 100cs$$

$$= \frac{5.4 \times 100}{60}cm = 9 \text{ cm}$$

Q. An exchange server 2000 subscriber. If the average BHCA (busy hour calls attempt) is 10,000 and the CCR (call completion rate) is 60%, calculate the busy hour calling rate.

Solution:

$$\text{Busy hour Calling rate} = \frac{\textit{average busy our calls}}{\textit{Total number of subscriber}}$$

$$\text{Average busy our call} = \text{BHCA* CCR}$$

$$= 10{,}000 \times \frac{60}{100}$$

$$= 6000$$

$$\text{Busy hour calling rate} = \frac{6000}{2000} = 3.$$

Q. Over a 20 minutes observation interval , 40 subscribers initiate calls. Total duration of calls is 4800 seconds. Calculate the load offered to the n/w by the subscriber and average subscriber traffic.

[note: offered traffic $A = C_{th}$]

Where,

C = average call arrival rate.

Th = average holding time per call.

C = 40/20 = 2.

th = 4800/(40×60) = 2 min per calls.

A = C.th

= 4E.

Q. During a busy hour 1400 calls were offered to a group of trunks and 14 calls were lost. The average call duration has three minutes find.

(1) Traffic offered (A)

(2) GOS (A_o)

(3) Traffic carried.

Solution:

$$\text{GOS} = \frac{\textit{Lost calls}}{\textit{Offered calls}} = \frac{14}{1400} = \ldots.$$

$$\text{Successful calls} = 1400 - 14$$

$$= 1386 \text{ calls.}$$

Traffic carried (A_o) = accepted no of calls * average holding time in hours .
= 1386*(3/60)
= ……… Erlangs or E.
Traffic offered (A) = offered calls * Average holding time in hours.
= 1400 * (3/60) = ………. E.

Q. An exchange is design to handel 2000 calls during the busy hour. One day, the number of calls during the busy hour is 2200 What is the resulting GOS.

Q. A call in processor in an exchange requires 120 ms to service a complete call. What is the BHCS rating for the processor? . If the exchange is capable of carrying 700 erlang of traffic, what is CCR?

Q. A total of 800 calls is offered to a switching system during the busy hour if 8 calls are lost due to insufficient equipment. What is the GOS? If the average holding time of successful call is 3 minutes, how much traffic is carried? What is the probability of loosign a call during a busy hours.

Q. During the busy hour group of ckt if offered 100 calls having an average duration of 3 minutes, and one call fails to find a free ckt. Calculate the traffic offered to the group, traffic carried by the group, traffic lost and the call conjection.
[Note: Time cojection = Total time duration which all devices/circuits were busy during a time interval]. Eg. $\frac{an\ hour}{The\ interval\ considered}$

Q. During the busy hour if all ckt in a group were simultaneously engage for a total period of 6 seconds calculate time conjection.